Belgacom
Forum™ 523/524

Mounting and Commissioning

Manual

belgacom

Welcome to Belgacom

Thank you for buying this Belgacom product. Our products meet the highest quality demands and are outstandingly designed.

The following instructions will guide you in the operation of your Forum™ 523/524 and answer all important questions. Should you require further assistance or information, please contact the person responsible for your system or your retailer first.

Internet:
www.belgacom.be/pabx

You can contact our Support on the following telephone numbers:

in Dutch: 0800 22 400

in French: 0800 33 400

in German: 0800 44 400

in English: 0800 55 400

Forum™ 5000 and Forum™ 500

This user guide applies to the Forum™ 5000 and Forum™ 500 product families. The Forum™ 500 product family comprises the Forum™ 523/524, Forum™ 525/526 and Forum™ 550/560 communications systems. The Forum™ 5050 communications system belongs to the Forum™ 5000 product family.

If individual features differ on the systems, a reference is made in this user guide.

We hope you enjoy using Forum™ 523/524.

Contents

About this Manual

This user guide is intended for the technician installing the Forum 523/524 and putting it into operation. This will usually be the responsibility of your Belgacom agent, who will prepare the Forum 523/524 and hand over the configured, operational system.

The information on

- installing the Forum 523/524,
- connecting terminals and putting them into operation, and
- installing and configuring the software of components

is intended for these persons only.

Please note: *The Forum 523/524 may be installed and serviced by qualified Belgacom personnel only!*

For users of the Forum 523/524, this manual provides useful information with regard to routine tasks and procedures (e.g. configuring teams). In addition, a list of frequently asked questions will help the user to detect and eliminate faults that may arise.

Features

Forum 523/524 is a communications system for integrated voice and data communication. It is an innovative, modular and convergent system platform designed primarily for VoIP voice and data applications. VoIP ("Voice over Internet Protocol") enables you to make telephone connections using Intranet and Internet data links, making it possible to use new applications, and providing cost advantages in many cases. The Forum 523/524 can also be used to make conventional telephone calls.

- The Forum 523/524 is equipped with all the interfaces necessary for connecting system telephones, IP system telephones, ISDN terminals and analogue terminals as part of its basic module. The system enables Internet-/Intranet data communications, CTI applications and system configuration using a standard web browser.
- Forum 523/524 has two slots for extra interface cards. Using different combinations of interface cards, the configuration of Forum 523/524 can be exactly tailored to your communication requirements. The need for an additional VoIP gateway, U_{pn} ports supporting DECT, further S_0 interfaces or more a/b ports can be met by using one or more interface cards without changing the system.
- The operating software is stored on a memory card (CompactFlash). This memory card can also be used to operate additional applications and programme packages: the digital voice memory and voice information systems Forum Voicemail and Forum Auto Attendant.
- A powerful Intel XScale CPU and implementation of the Linux operating system guarantee the security of your investment, compatibility with current Internet applications and future expandability.
- An integrated Ethernet Switch also enables the use of applications featuring media convergence, such as the use of a Media Gateway card for VoIP applications.
- The Forum 523/524 also has an actor for a door opener.

Telephony

The Forum 523/524 communications system is designed to be connected to an ISDN-basic access using the DSS1 protocol or to be connected using Internet telephony according to RFC 3261 via a SIP provider. System access (point-to-point) and multi-terminal access (point-to-multipoint) are both supported. All types of access can be configured in parallel. Forum 523/524 complies with legal regulations governing telecommunications equipment and fulfils the DSS1protocol.

From its smallest version, Forum 523/524 provides the following connection options on its basic module:

- 1 external S_0 port, provided as an RJ45 socket
- 1 switchable S_0 port, provided as an RJ45 socket (external) and as a 4-pin pressure terminal (internal)
- 3 DECT-enabled U_{pn}-connections for system telephones or DECT base stations (pressure terminals)
- 4 a/b-connections for analogue devices (pressure terminals)
- Forum 523/524 can be expanded using an appropriate interface card to 4 S_0 connections, 8 U_{pn} connections or 8 a/b connections. You will find an overview of the cards available in the section *Interface Cards* starting on page 40.

You can connect the following devices to Forum 523/524:

- analogue terminals to a/b ports
- Euro ISDN terminals in accordance with DSS1to internal S_0 ports
- Belgacom system telephones Forum Phone 515, Forum Phone 525, Forum Phone 535 and Belgacom system telephones Forum Phone 520, Forum Phone 530 to U_{pn} ports
- VoIP system telephones Forum IPhone 525 and Forum IPhone 535 as well as Forum IPhone 520 and Forum IPhone 530, can be connected using LAN and an additional Media Gateway card.
- Base stations for DECT mobile phones to DECT-enabled U_{pn} ports

- DECT over IP base stations for DECT handsets can be connected using LAN and an additional Media Gateway card.

If your network provider supports CNIP (Calling Name Identification Presentation), this can show you the names and numbers of callers for each incoming call. Forum 523/524 supports the display of these names on system telephones. If however, you have created an entry under the caller's number in the telephone book Forum 523/524, this will be displayed instead.

Forum 523/524 can be integrated into an existing network (LAN) and be used from all workstations as Internet access router. VoIP integration and the connection to an existing EMail system is possible.

The Forum 523/524 is configured and programmed using a special web browser (the Web console), which can be run on a connected PC.

The configuration of the Forum 523/524 can be prepared by a Belgacom agent. The Forum 523/524 configuration can also be edited or updated by the Belgacom Customer Service Centre via remote access.

To connect Forum 523/524 to existing company hardware, an actor can be connected to a door opener contact. A doorbell and door intercom can also be connected using additional devices.

Forum 523/524 enables you to use CTI (**C**omputer **T**elephony **I**ntegration) applications. The TAPI (Telephony Application Programming Interface) and CSTA (Services for Computer Supported Telecommunications Applications) standards are supported for the purpose of integrating CTI applications.

Forum 523/524 also features an integrated browser-based CTI application, the Forum CTI. The Forum CTI allow users to call up and use telephone functions from their PCs.

Furthermore, the Forum CTI Touch web application is a web-based solution available for controlling telephony functions with current smartphones.

Further Telephony Features

The installed memory card enables you to operate a digital voice memory and voice information system. A 1 GB memory card offers about 60 minutes of recording time. For more information, please refer to the user guides “Forum Voicemail” and “Forum Auto Attendant”.

You can optimize your telephone communications by using team functions and call-queuing function.

The separate licensable Forum Count Web application enables you to record, store and analyse telephone connections using configurable filters. You will find more information on this topic in the Online Help of the Web console.

You can integrate external telephony applications via the extended CSTA interface. The extended CSTA interface is licensable separately (see *CSTA interface* starting on page 207).

Networking

As your company’s requirements grow, Forum 523/524 can be networked with other communications systems via permanent ISDN lines (“Q.SIG”) or via Internet connections (“Q.SIG-IP”), connecting distant locations and branch offices in a single, comprehensive telephone network. Forum 523/524 can for example, operate as a sub-system or a DECT server via the Q.SIG network. This type of networking is described in the chapter on *PBX Networking* starting on page 152.

DECT Networks

Forum 523/524 supports the setting up of a DECT network, enabling mobile office communications. For larger offices, using several DECT stations (RFP, Radio Fixed Parts) can expand wireless network coverage and provide a transparent handover between the RFPs while maintaining the telephone connection.

Handsets as system terminals with all the system telephone features are available for DECT telephony and handsets using GAP and CAP standards are also supported. The handover between the RFPs also functions with handsets using the GAP standard.

Up to 4 calls can be made simultaneously via an RPF, and up to 8 calls can be made simultaneously if an RFP is connected via 2 U_{pn} connections. The DECT network can also be operated using VoIP (see *DECT over IP®* starting on page 140).

Voice over IP (VoIP)

Forum 523/524 supports the connection of VoIP terminals, enabling the company's existing internal network infrastructure (LAN with 100 MBit/s) to be used for telephony as well. Corded system terminals of the Forum IPhone 525 and Forum IPhone 535 as well as the VoIP system telephones Forum IPhone 520 and Forum IPhone 530 types are available for this purpose. These devices have the same functionality and support the same features as the non IP-enabled system terminals.

The eight gateway channels of an installable Media Gateway card are automatically switched on for telephone calls between IP terminals and ordinary terminals, for data compression, for the generation of DTMF and dial tones and for echo suppression. It is also possible to use a limited range of VoIP functions without an additional Media Gateway card. The system software provides up to 32 uncompressed VoIP gateway channels without echo compensation.

For users wanting to use PC-supported telephony, IP system terminals are available as separate licensable software versions ("Softphone"). You will find further information on this in the chapter on *Voice over IP (VoIP)* starting on page 101.

VoIP with media gateway card

Additional VoIP features can be used by installing a media gateway card with the communications system Forum 523/524.

- **SIP (internal)**: You can operate SIP system phones Forum IPhone 512 / Forum IPhone 545 and standard SIP telephones.
- **SIP (external)**: You can use external SIP connections as "SIP trunk lines". Setting up and using external SIP connections is completely transparent for telephone users, providing them with easy access to low-cost Internet telephony and a fallback to normal ISDN connections in case of error or busy lines. A Media Gateway card is required for SIP telephony.

- **Q.SIG-IP** and **SIP tie line**: Several communications systems can be networked via IP connections using “Q.SIG IP”. Low-cost data connections can be used to network the communications systems of branch offices instead of ordinary permanent ISDN lines.
- **DECT over IP®**: DECT networking via VoIP is another possible option for offices already extensively using VoIP telephony. The Radio Fixed Parts (RFPs) are connected via network data connections, so they do not occupy any U_{pn} ports and can use existing network connections. With DECT over IP, VoIP protocol data is changed into DECT-compatible voice data direct on the RFPs. DECT-RFPs and DECT over IP-RFP can be used together in combination in many cases; it is however not possible to switch between RFPs using different technologies during a call.

Packet Data in the D Channel

Some business applications, for instance POS terminals, cash registers or credit card terminals, require a permanent data connection over the X.25 packet data network. Packet data transfer through the ISDN D channel (according to X.31 via SAPI 16) can also be established between several S_0 interfaces of the Forum 523/524. Simultaneous connections are distinguished by means of a TEI (Terminal Endpoint Identifier).

X.31 packet data can be forwarded between two S_0 interfaces (for instance an internal and external S_0 interface). Equally, data can be forwarded (“routed”) over permanent Q.SIG lines. It is possible to operate multiple terminals with the same TEI on different internal S_0 interfaces. A TEI mapping table allows these X.31 connections to be routed to the same external S_0 interface.

The routing table for X.31 packet data is set in the Configurator under **Telephony**: **Extended**: **X.31**. Additional information can be found in the Configurator online help files.

Internet Access

It is possible to connect a complete company network (LAN, Local Area Network) with the Forum 523/524. All work station computers within the LAN can then access the Internet via the Forum 523/524. If Internet access is already available from an Internet Service Provider, this can also be configured in the Forum 523/524. The Forum 523/524 provides the IP configuration required for Internet access. A DHCP server and a DNS server are integrated into the Forum 523/524 and these handle the administration of IP addresses and name resolution for the client computers.

The Forum 523/524 provides Internet access for all connected PCs through a common IP address. Only this address is externally visible in the Internet. Network Address Translation (NAT) translates the local IP addresses of the client computers to the IP address of the Forum 523/524. This means that the workplace computers in the LAN cannot be directly reached from the outside (i.e. from the Internet) and are thus protected from direct attack from the Internet. The LAN is additionally protected by the Forum 523/524 filter lists, which can be individually configured (Firewall function).

Note: *We recommend you read the explanations in Useful Information on Internet Access starting on page 98.*

E-Mail

An e-mail function, which can use the POP3, APOP or IMAP4 protocols to check the Internet Service Provider for incoming mail, is integrated into the Forum 523/524. Mail account retrieval can be configured for each member of staff when configuring Forum 523/524.
Forum 523/524 then fetches the headers (Subject) and sender information from incoming e-mails from the mail servers at preset intervals and forwards them to the user's system terminal.

E-mail accounts for sending mail can also be configured for users. E-mails can then, for example, be sent direct from the Forum CTI to other users. In addition, users who have had a voicebox configured for themselves can be notified by e-mail of new voicebox messages.

Forum 523/524 records important events and errors in an internal logbook, the error store. Entries in the logbook (system messages) can also be sent by e-mail to inform or alert system administrators.

Further Network Features

You can also offer your staff the option of dialling into the LAN by means of RAS access. This type of connection can also be made using an encrypted Internet data connection (VPN, Virtual Private Network).

Branch offices can also be linked via ISDN or VPN. This means that two Forum 523/524 can connect their LANs either by dialling in on demand or using an encrypted Internet data connection.

You can query the internal phone book of the Forum 523/524 communications system via LDAP. It is also possible to integrate entries of an external directory via LDAP. Furthermore, integration into a network administration solution via SNMP is possible. For further information on the various application interfaces, please refer to *Application Interfaces* starting on page 207.

Glossary

Please refer to the explanations in the Glossary (supplied as a PDF file on the system CD).

Factory Settings on Delivery

The following basic settings and features are active on delivery. We recommend that you configure the Forum 523/524 communications system to suit your individual requirements before using it (see *Configuration* starting on page 72).

Note: *The basic settings apply to the smallest version of Forum 523/524, i.e., without additional interface cards.*

Telephony Basic Settings

- The $S_0$1 port is configured for multi-terminal access, the $S_0$2 port for system access.
- The 3 U_{pn}-ports are configured for Forum Phone 525 system telephones with the call numbers 30 to 32.
- The 4 a/b ports are configured for analogue devices with the call numbers 10 to 13.
- The Forum 523/524 is configured ready for use in Belgium.
- All corded terminals ring if there are incoming external calls.
- The password for the “Administrator” user is empty.
- The system PIN, for remote programmable call diversion for example, is set at “0000”.

Authorisations

The allocation of authorities regulates which functions terminals on Forum 523/524 can use. These authorities are configured for so-called user groups, to which users and their terminals are then assigned.

Three user groups are preset: “Administrators”, “Standard” and “Guests”. “Administrators” can access all functions in Forum 523/524 and have unlimited configuration rights. Users in the “Guests” group can not configure Forum 523/524, may not make any external calls and can only use a limited range of Forum 523/524 terminal functions. The “Standard” user group is well suited as a starting point for creating user groups for normal users of the system (e. g. the staff members of a company) because of its default settings.

Note: *When the Forum 523/524 is commissioned, all connected terminals are initially in the “Administrators” group until a user logs on to the Web console. Subsequently, all terminals are automatically in the “Guests” group. For more details on the configuration of user groups, refer to the online help in the chapter entitled “User Manager”.*

The following functions are delivered preset for user groups:

User group settings

Function / Authorisation	Standard	Administrators	Guests
Applications			
Configurator	personal	Expert	View
Costs	-	+	-
Phone Book	+	+	+
Forum CTI	+	+	-
Busy lamps	+	+	-
Forum Count	+	+	-
Forum Hotel	+	+	-
ISP application	-	-	-

User group settings

Function / Authorisation	**Standard**	**Adminis-trators**	**Guests**
Courtesy Service	off	off	off
Phone Book			
Use LDAP	+	+	+
Use central	+	+	+
Edit central	-	+	-
Use own company	+	+	+
Edit own company	-	-	-
Edit other companies	-	-	-
Entries (personal)	20	20	0
Dial in (outgoing)			
External	International	International	Incoming only
Immediate external line seizure	-	-	-
External line seizure over operator	-	-	-
LCR *)	+	+	-
Deactivate LCR *)	+	+	-
LCR at call forwarding to extern. *)	-	-	-
VIP call *)	+	+	-
PIN dial *)	-	-	-
Announcement *)	+	+	-
Announcement accept *)	+	+	+
Intercom *)	+	+	-
Dialout for other phone *)	-	-	-
Instant connection*)	+	+	-

User group settings

Function / Authorisation	**Standard**	**Adminis-trators**	**Guests**
Callback on busy *)	+	+	-
Multiple seizure at the parallel terminal s*)	+	+	+
Switch authorization *)	-	-	-
Display phone number off (intern) *)	-	-	-
Display phone number off (extern) *)	-	-	-
Display phone number off/ on per connection *)	+	+	-
Dial in (incoming)			
Pickup from group	+	+	-
Pickup selective	+	+	-
Take	+	+	+
Call removal *)	-	-	-
Calling suppression at the parallel terminal *)	-	-	-
Reaction: connection will be disconnected *)	-	-	-
display phone number off (intern) *)	-	-	-
Call queue *)	0	0	0
Call forwarding			
Call forwarding	+	+	-
Call forwarding to extern	+	+	-
Call forwarding of MSN groups	+	+	-
Call forwarding door call	+	+	-

User group settings

Function / Authorisation	Standard	Adminis-trators	Guests
Indicate call forwarding after time parallel *)	+	+	+
Call forwarding for other user *)	-	-	-
Prevent call forwarding by other user*)	-	-	-
Display: Call forwarding via *)	last forwarding	last forwarding	last forwardi ng
Connection *)			
External to external *)	+	+	-
3-party conference *)	+	+	-
Park call *)	+	+	-
MOH at external connections*)	+	+	+
MOH at internal connections *)	+	+	+
Protection			
Call protection	ringing tone	ringing tone	off
Call waiting protection	+	+	-
Announcement protection *)	-	-	-
Intercom protection *)	-	-	-
Pickup protection *)	-	-	-
Phone lock *)	+	+	-
Intercept *)	+	+	-
Lists			
Black lists	empty	empty	empty

User group settings

Function / Authorisation	**Standard**	**Adminis-trators**	**Guests**
White lists	empty	empty	empty
Special lists	1	1	1
Call filter	empty	empty	empty
Manage intern call list *)	+	+	-
Manage extern call list *)	+	+	-
Manage busy call list *)	+	+	-
Manage door call lists *)	+	+	-
System phones *)			
Device busy *)	+	+	+
All keys locked *)	-	-	+
Programming function keys *)	+	+	+
Menu and ABC keys *)	+	+	+
DECT trunc keys *)	-	-	-
Disconnect ISP connection *)	+	+	-
Connection data *)			
Send incoming connections *)	-	-	-
Send outcoming connections *)	-	-	-
Recording incoming connections *)	-	-	-
Recording outgoing connection *)	-	-	-
Number of suppressed digits *)	0	0	0

User group settings

Function / Authorisation	**Standard**	**Adminis-trators**	**Guests**
Incoming basic amount *)	0,00	0,00	0,00
Outgoing basic amount *)	0,00	0,00	0,00
Cost factor *)	100%	100%	100%
Create costs *)	-	-	-
Network *)			
RAS *)	-	-	-
Callback *)	none	none	none
E-mail notification *)	+	+	-
Send E-mails*)	+	+	-
CSTA *)			
CSTA menu *)	-	-	-
CSTA key *)	-	-	-
CSTA data *)	-	-	-
Other *)			
Speed dialling *)	+	+	-
Door opener *)	+	+	-
Keypad dialling *)	+	+	-
Time control *)	-	-	-
SMS stationary *)	-	-	-
Booking number may be set up*)	+	+	-
Send short messages*)	+	+	-

*) These settings are shown only in the Expert view.

The following important settings are active without further configuration:

- External authorisation: International numbers can be dialled from all configured terminals. External lines must be seized by entering a preset code.
- Call forwarding to internal and external numbers can be activated. Call forwarding after delay is performed after 20 seconds. Door calls and MSN groups can be forwarded. Call diversions for other users and call diversions by other users are deactivated.
- The telephone lock can be activated. The terminal PIN is “0000”.
- The white list, black list and call filter are not preconfigured and therefore not active. If these lists are configured, they can be activated for user groups. A special list of emergency phone numbers is preset and activated.
- The door opener can be opened from all terminals. Door calls can be forwarded.
- Every standard user can change the configuration of Forum 523/524.
- Every standard user can create a personal telephone book and edit entries in the central telephone book.
- Every standard user can read out the charges.
- Applications requiring a licence (e.g. Forum Count) can be used after being activated.
- RAS access is not allowed.

Internet Functions

- RAS access can be configured for every user of Forum 523/ 524. RAS access enables remote users to dial in via VPN (PPTP or IPSec) or via ISDN (with or without call-back). RAS access requires the activation of the RAS authorisation.
- Multiple e-mail account queries can be set up for every user. Every user with a system terminal can be automatically informed about incoming e-mails with this function. Subject and sender of the e-mail will be displayed. The e-mail remains on the e-mail-server where it can be queried with a standard e-mail programme.
- Users can disconnect existing Internet connections (via the Web console of Forum 523/524 and from a system terminal, if this function has been configured on this terminal).

The following IP addresses are preset for the network configuration:

- Host Name: host
- Domain Name: domain
- IP address: 192.168.99.254
- Network mask: 255.255.255.0

The following addresses are assigned to the client computers by DHCP or PPP:

- Gateway address: 192.168.99.254
- Domain Name: domain
- Domain Name Server: 192.168.99.254
- PPP addresses (RAS):
  ISDN: 192.168.99.10 to 192.168.99.41
  PPTP: 192.168.99.50 to 192.168.99.79
  IPSEC: 192.168.99.90 to 192.168.99.119
- DHCP addresses (LAN): 192.168.99.130 to 192.168.99.169

You can change the IP settings in the **Configurator**. Talk to the system administrator responsible for the existing LA N if you wish to do this.

Update Notes

If you are already operating an Forum 523/524 communications system with an older version of the software, the following notes will help you update to the release 7.

The release 7 is published at the same time for all members of the Forum 500 / Forum 5000 product family. If you are operating multiple communications systems, for example, on TC system networking, updating all communications systems at the same time is a good idea to ensure optimal compatibility.

Upgrade Licence

You can purchase a new Forum 524 communications system. This offers an extended range of functions as opposed to the Forum 523 prior model.

New features Forum 524

Feature	Explanation
NTP	Time synchronisation via internet
SNMP	network management integration
SIP tie-line	TC system networking via IP
transparent codecs	free SIP terminal codec negotiation (e.g. video)
LDAP client	integration in existing address data
Forum Hotel	extended hotel function
Forum Auto Attendant	extended attendant function
Forum CTI Touch	CTI solution for smartphones

You can also use these features as needed on an existing Forum 523 communications system. You can thus purchase an upgrade licence:

- When updating the software to release 7 you can use the new release 7 features without the extended range of functions.

- An upgrade licence enables you to use the new release 7 features including the extended functions (see table *New features Forum 524*). When you enter the upgrade licence authorisation key, the communications system changes its designation to Forum 524.

The upgrade licence is already included when you purchase a new Forum 524 communications system.

Technical Notes

The extended release 7 features may require you to upgrade the existing communications system. Please note the following points:

- A 256 MByte CompactFlash card is the prerequisite for updating to release 7.
- The system software is divided into 2 parts: a software container with the firmware which you load into the communications system when you update. And a software container ("firmware extensions" for certain terminal types) which you only load after updating the communications system.
- The firmware extensions contains software for the following terminal types:

Type	**Model**
SIP system phones	Forum IPhone 512 Forum IPhone 545
DECT over IP base stations	Forum Base DECT IP Forum Base DECT IP v2

Note: *If you are not operating any of these terminals, you do not have to load the firmware extensions.*

Updating

Updating is done in the following steps:

1. Open the **System**: **Firmware** page in the **Configurator**. Click on the **Save** button to create a backup copy of the existing configuration. Then click on the **Next** button.
2. Click on the **Browse** button. Select the firmware file. Click on the **Load** button.

   This process loads the firmware file to the communications system. After loading, the communications system re-starts. Then the existing configuration is converted. This process can take up to 20 minutes depending on how much configuration data is involved.
3. Then open the **System**: **Components** page. Select the **Firmware Addons** entry from the selections offered. Take note of the expected file name displayed. Load the firmware extensions with the **Browse** and **Load** buttons.
4. Optional: Then open the **System**: **Licences** page. Enter the authorisation key for the update. Then a green tick mark should appear next to the Forum Upgrade entry in the **Status** column.

After updating, as needed, you can also load the online help file and the audio files for the additional programs Forum Voicemail und Forum Auto Attendant.

If you have been operating external SIP lines with customer-specific settings, you may have to adjust these settings to the extended SIP provider settings of release 7 (see online help topics “SIP Trunks”, “SIP Provider” and “Notes on SIP Providers”).

Note: *Please also note the information on updating telephony functions (see Information on the Update starting on page 173).*

Installation

To install the Forum 523/524 please perform the following steps in the order suggested.

1. Check the scope of delivery (see *Scope of Delivery* on page 32), read *Safety Precautions* starting on page 32 and information on *Mounting Location* starting on page 35.
2. Open Forum 523/524 (see *Opening and Closing Forum 523/524* starting on page 36)
3. Install interface cards, if provided (see *Interface Cards* starting on page 40)
4. Mount the device on the wall (*Mounting* starting on page 47)
5. Connect terminals (see *Port Assignment, Termination, Cable Length* starting on page 48)
6. If available: externally connect S_0 to NTBA (see *S_0 Ports* starting on page 48)
7. Plug power cord into power supply unit
8. Close Forum 523/524 (see *Opening and Closing Forum 523/524* starting on page 36)
9. Switch on power supply

   Wait until the system is initiated (takes about 2 minutes). The system telephone software is initialised automatically (the display will then show the telephone’s internal phone number).

   Call an external number to check basic functionality.
10. Connect PC/network to LAN1 for configuration
11. Close Forum 523/524
12. Open configuration software (Configurator) in the browser (see *Configuration* starting on page 72); access data are:

    Address (URL): http://192.168.99.254/
    User: Administrator
    Password: No password assigned
    System PIN: 0000

13. Configure the Forum 523/524 using the Configuration Assistant

14. *Loading SW Updates* (see page 83) and if applicable *Resetting the System Data* (see page 83)

Scope of Delivery

The delivery consists of:

- Forum 523/524 communications system in its basic configuration (with basic module)
- 1 power supply unit with mains power cable
- Connection cable for the ISDN S_0 port
- One set of mounting screws and dowels
- 1 CD containing complete documentation and software

Safety Precautions

The CE symbol on the product confirms that it conforms with the technical guidelines on user safety and electromagnetic compatibility at the time of its approval.

Intended Usage

The Forum 523/524 communications system is designed to be connected to ISDN basic rate access (“multi-terminal access or “system access) using the Euro-ISDN protocol (DSS1). The Euro-ISDN basic rate access must be equipped with the Network Termination Basic Access (NTBA) of your network operator.

The Forum 523/524 is not designed to be used with other telecommunications access. Incorrect usage can lead to malfunction or damage to the Forum 523/524 and to the network.

Installation

The Forum 523/524 may only be installed inside buildings and may only be operated when mounted on a wall.

Do not install the Forum 523/524 during a storm. Do not connect and disconnect lines during a storm.

CAUTION!

Static charges can damage the Forum 523/524's electronic components. Please make sure that you discharge yourself and your tools before and during any installation work on the Forum 523/524.

Connection to the Mains Power Supply

The Forum 523/524 may only be plugged into mains power sockets which have a protective earth conductor. It is not necessary to provide any additional earthing for the Forum 523/524.

Recommendation: Connect the Forum 523/524 to a separate 230 V power circuit so that short circuts occurring in other devices do not put the Forum 523/524 out of operation. The mains power connection must be installed by a licenced electrician to avoid danger to people or materials!

DANGER! Dangerous voltage in device! Pull the plug out of the power socket before opening the device and connecting any lines/terminals or installing interface cards! Failure to do so may cause danger to life from electric shock!

The Forum 523/524 does not have its own power supply switch. To disconnect the Forum 523/524 from the mains power supply, pull the plug out of the power socket.

Install the Forum 523/524 near an easily-accessible mains power socket so that the plug can be quickly pulled out of the power socket in a hazardous situation.

Power Supply Unit and Power Cord

Use only the power supply unit and power cord provided to connect the Forum 523/524 to the mains power supply.Other power supply units and power cords may cause malfunctions or electric shock and damage the Forum 523/524.

CAUTION! Never start or operate the Forum 523/524 if the power supply unit or power cord is damaged. Serious danger to life from electric shock may result.

Cables and Terminals

Ensure that all cables are laid in such a way that nobody can walk on or trip over them.

Only U_{pn} lines, a/b lines and switching circuits (door opener to actor) may be installed outside buildings. In this case, **no** S_0 terminals may be operated. Outside cables and terminals may only be installed and connected to the Forum 523/524 by qualified specialists observing the appropriate safety measures.

Forum 523/524 and all connected terminals must be electrically disconnected from the earth ground (protective earthing). For example: where you have an earthed door, make sure that the door opener connected to the actor port of the Forum 523/524 does not come into contact with the metal parts of the door.

Only devices which provide SELV (Separated Extra Low Voltage) may be connected to Forum 523/524. Proper use of authorised devices complies with this stipulation

Only terminals which fulfil the technical requirements may be connected to the analogue ports. Please see the chapter on *a/b Ports* starting on page 55 for further information.

Use a shielded Ethernet cable (STP cable, Shielded Twisted Pair cable) to connect Forum 523/524 to a local network (LAN, Local Area Network).

Usage

Make sure no fluids to get into the Forum 523/524: electric shock or short circuit may result.

Repairs to the Forum 523/524, and all its accessories and terminals must be carried out by accredited specialists. Inappropriate repairs may damage the Forum 523/524 and render any warranty claims invalid.

Keep the Forum 523/524 and its accessories and packaging out of reach of children!

Mounting Location

The ambient temperature for operating the Forum 523/524 must be between +5 °C and +40 °C. The power supply must be 230 V/ 50 Hz AC and using a separate fuse for the power supply is recommended.

To maintain the prescribed ambient temperature, mount the Forum 523/524 in a well-ventilated location away from direct sources of heat.

Mount the Forum 523/524 on a wall, but:

- not in front of or above heat sources such as radiators,
- not in places subject to direct sunlight,
- not behind curtains,
- not in small, unventilated, damp rooms,
- not on or near inflammable materials
- and not near high-frequency devices such as transmitters, x-ray equipment or any similar devices.

Connect the device to a separate 230 V power circuit and install a mains overload filter.

Opening and Closing Forum 523/524

Forum 523/524 is designed to be installed by the user. Live parts should not be accessible to users or if they are accessible to users, should be accessible only to a user using the appropriate tool. For this reason, Forum 523/524 features a special casing mechanism, which can only be opened step by step.For your own safety, please read the *Safety Precautions* starting on page 32. As with all highly sensitive electronic devices, static charges can destroy components in a fraction of a second. For this reason, you should first pick up your tool then touch an earthed metal object (e.g. a radiator), before touching any connections or components inside the device.

CAUTION!

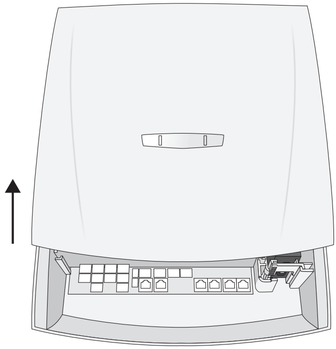
Static charges can damage the electronic components of the Forum 523/524.

There are no directly accessible circuits operating under dangerous voltage on Forum 523/524's connection panel. You must still switch off all power to the device when performing any kind of installation. Pull the plug out of the mains power supply socket.

It is recommend to connect the pressure terminals provided here as sockets to externally mounted sockets – e.g. UAE sockets for U_{pn} ports, IAE sockets for S_0 ports and standard telephone sockets for a/b ports. You can plug these connecting sockets into terminals and do not have to switch off the device to do this.

To check the internal Light Emitting Diodes (LEDs) or connect telephone sockets, terminals or network connections, open the Forum 523/524 as far as the plate:

1. Pull the plug out of the power socket in the wall.



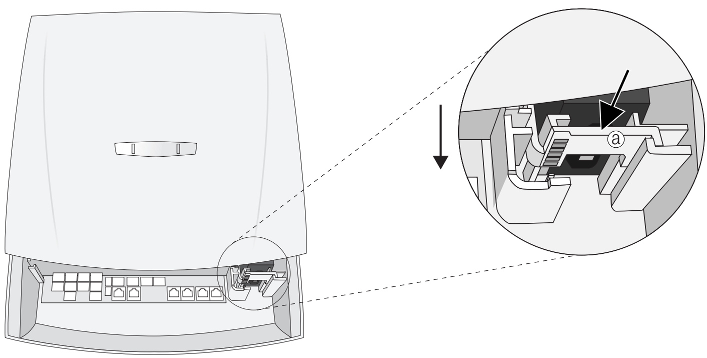
Push up the cover

2. Push the cover open using light pressure until the connector panel can be seen. A stopper prevents the cover from being opened further.

   You must remove Forum 523/524's cover completely if you want to install it on a wall or install interface cards:

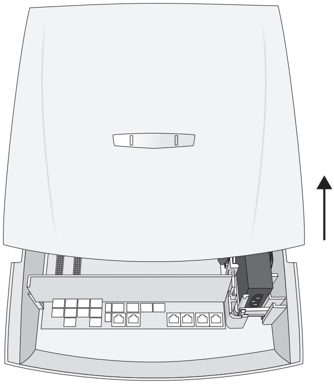
3. Pull the 3-pin AC cable out of the power supply. unit

4. Press down the power supply locking device.



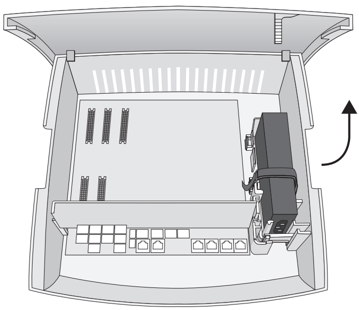
Pressing down the power supply locking device

5. It is only possible to open the cover further after the locking device has been tripped. Push the cover back to approx. 5cm beyond the protection plate.



Pushing the cover further back

6. Take the cover off from the bottom.



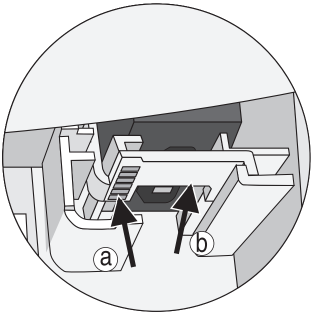
Taking the cover off

To close the Forum 523/524 again, follow this procedure in reverse order:

7. Put the cover back on.

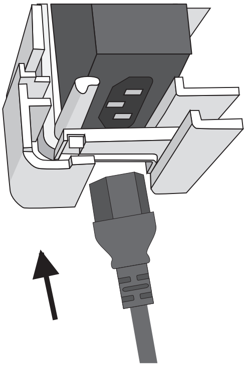
8. Push the cover down as far as the protection plate

9. Lift the stop lever with an appropriate tool where necessary (a). Press the power supply locking device up again (b).



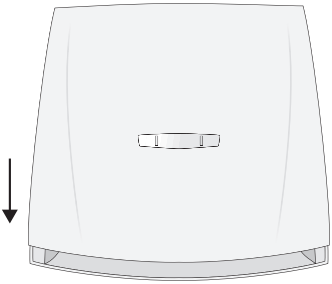
Locking in the power supply unit

10. Connect the 3-pin AC cable to the power supply unit.



Connecting the 3-pin AC cable

11. Push the cover all the way down



Close the cover

Interface Cards

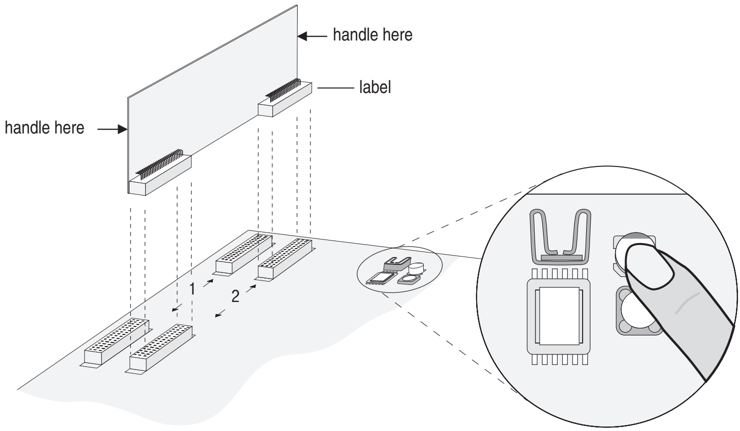
Forum 523/524 can be expanded through the use of interface cards. Forum 523/524 has two slots for interface cards.

Please note: *Switch off the Forum 523/524. Pull the plug out of the mains power supply socket. Never install or de-install interface cards when the Forum 523/524 is switched on.*

Installing Interface Cards

Each interface card is connected to the main board via two sockets. Forum 523/524's slots have the following characteristics:

- There is no prescribed order for assignment. You can, for example, use an interface card in slot 2, although slot 1 is not assigned.
- Slot 1 is connected to a group of pressure terminals. The pressure terminals of the slot are given the same colour to make them easier to identify.
- The slots are not identical. Interface cards can therefore not be used in all and any slots. Please see the overview under *Position of the Interfaces* starting on page 42 for details.



Installing an interface card

To install an interface card, please follow these steps:

1. Switch off the Forum 523/524. Open the device (see *Opening and Closing Forum 523/524* starting on page 36).
2. To discharge, touch a radiator or another metallic installation connected to earth ground. Take the interface card out of its packaging. Check that it is the correct type of card. There is a label specifying the type on the connector.

CAUTION!

Static charges can damage the Forum 523/524's electronic components, so you must provide potential equalisation between you and the system. Hold the edge of the interface card with one hand. With the other hand briefly touch the capacitor on the board.

3. Carefully plug the interface card into its designated slot. The label on the interface card's connector must be facing to the right.

   Make sure it is sitting firmly in the slot.
4. Close the cover as far as the protection plate (see *Opening and Closing Forum 523/524* starting on page 36, steps 7 to 9). Connect the required connection cables to the relevant connection blocks (see also *Position of the Interfaces* starting on page 42).
5. Close the cover of the device completely. Switch on the Forum 523/524.

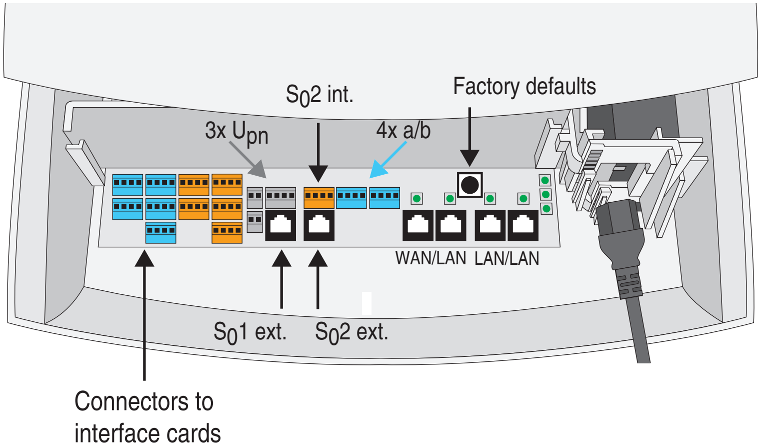
Forum 523/524's software can identify the type of interface card. After installation the interface card must be individually configured before starting operations.

Once you have restarted Forum 523/524 you can view the interface cards' status in der Web console. to do this, call up the menu item **Telephony: Ports: Slots**. You will see a green tick in the **Status** column in the table next to the slot's number (**1** or **2**). The type of interface card must also be also shown in the **hidden** column.

Position of the Interfaces

The following overview shows the available interface cards and the assignment of the relevant connection blocks.

The following diagram shows the position of the interfaces:

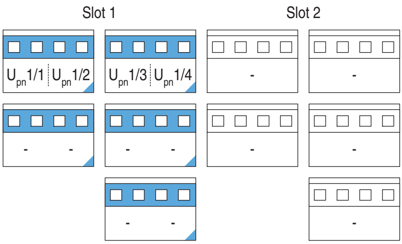


Position of the Forum 523/524 interfaces

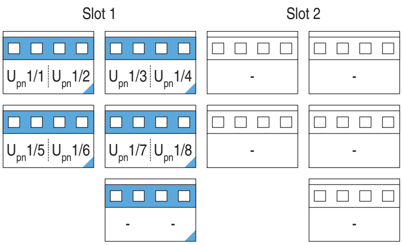
The following overview shows the interface cards available.

Interface card	Slot 1	Slot 2	Features
Forum 523 / 525 Media Gateway Card		●	Internal connection to Ethernet switch via the slot
Forum 523 / 525 Card 4 digital extensions: 4 x U_{pn}	●		U_{pn} also usable for DECT-RFP
Forum 523 / 525 Card 8 digital extensions: 8 x U_{pn}	●		U_{pn} also usable for DECT-RFP
Forum 523 / 525 Card 2 S0/T0, 6 digital extensions: 2 x S_0 and 6 x U_{pn}	●		U_{pn} also usable for DECT-RFP S_0 are internally/ externally switchable
Forum 523 / 525 Card 2 S0/T0, 6 analogue extensions: 2 x S_0 and 6 x a/b	●		S_0 are internally/ externally switchable
Forum 523 / 525 Card 4 S0/T0: 4 x S_0	●		S_0 are internally/ externally switchable
Forum 523 / 525 Card 4 analogue extensions: 4 x a/b	●		
Forum 523 / 525 Card 8 analogue extensions: 8 x a/b	●		

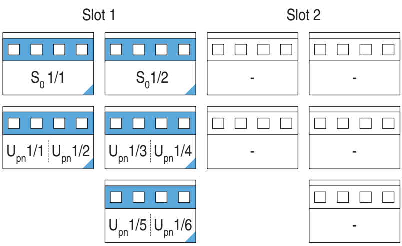
The following pressure terminal diagrams show the assignment of pressure terminals for one type of card only. If you use different types of card at the same time, the actual assignment will be a combination of several of these diagrams.



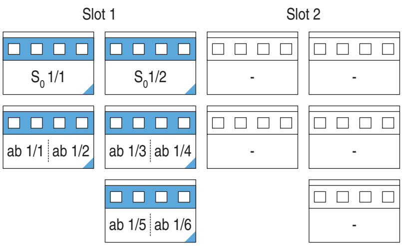
Forum 523 / 525 Card 4 digital extensions: 4 x U_{pn}



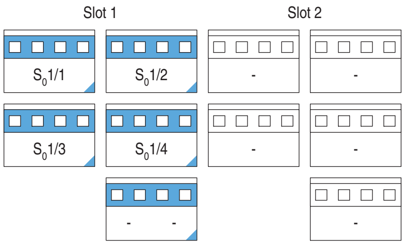
Forum 523 / 525 Card 8 digital extensions: 8 x U_{pn}



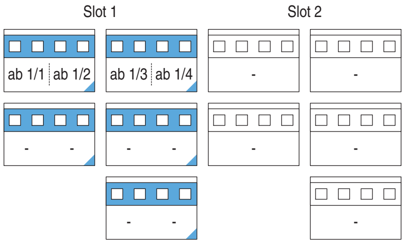
Forum 523 / 525 Card 2 S0/T0, 6 digital extensions: 2 x S_0 and 6 x U_{pn}



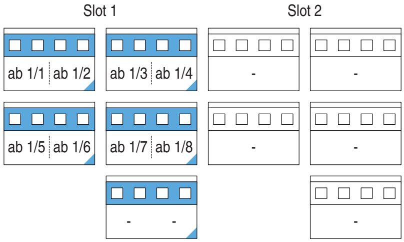
Forum 523 / 525 Card 2 S0/T0, 6 analogue extensions: 2 x S_0 and 6 x a/b



Forum 523 / 525 Card 4 S0/T0: 4 x S_0

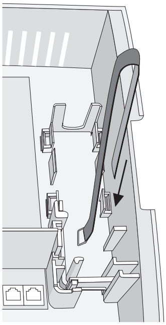


Forum 523 / 525 Card 4 analogue extensions: 4 x a/b

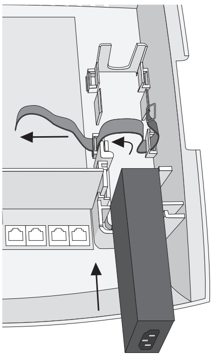


Forum 523 / 525 Card 8 analogue extensions: 8 x a/b

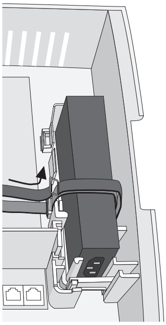
Installing the Power Supply Unit



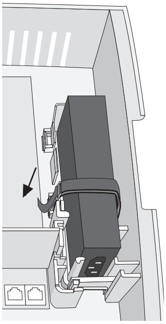
1. If you want to change the power supply unit, first switch off the Forum 523/524 and pull the plug out of the mains power supply.
2. Take the cover off the Forum 523/524 (see *Opening and Closing Forum 523/ 524* starting on page 36).
3. Pass the Velcro fastener through the recesses in the bottom of the casing.



4. Put the power supply unit into the mounting recess on the right, passing the Velcro fastener over the power supply unit.



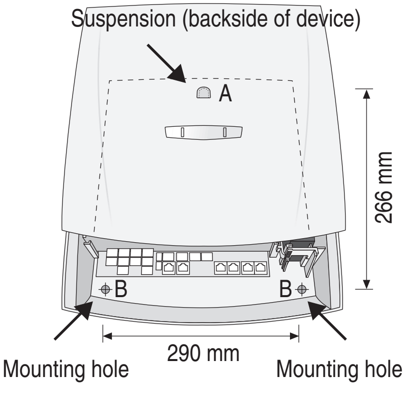
5. Pass the Velcro fastener through the lug.



6. Tighten the Velcro fastener.
7. Replace the cover and connect the power cable to the power supply unit (see *Opening and Closing Forum 523/ 524* starting on page 36).

Mounting

The device must be mounted in a suitable place. Please see the section on *Mounting Location* starting on page 35 for details. The Forum 523/524 is attached to the wall by means of 3 screws as shown in the diagram below:



Attachment diagram

Take the cover off the Forum 523/524 to screw in the screws for attchment point B and insert the screws through the holes provided. The Forum 523/524 will hang on the screws in attachment point A, so you must leave a space of 4 mm between the screw heads and the wall.

Port Assignment, Termination, Cable Length

S_0 Ports

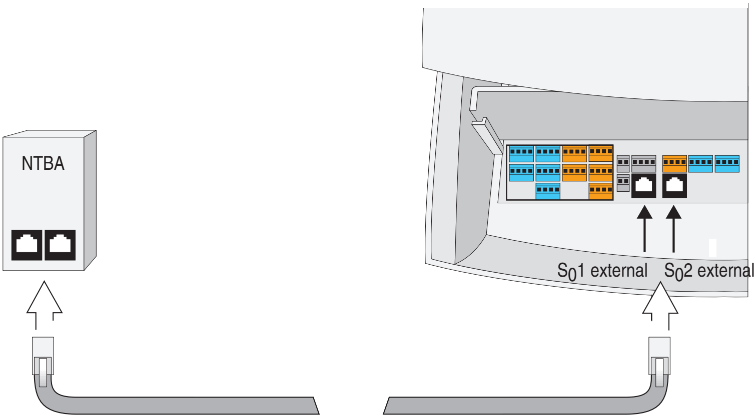
All S_0 ports can be externally connected (i. e. on ISDN network terminations). The $S_0$2 port and all S_0 ports on additionally installed interface cards can also be internally connected. Simultaneous use of one S_0 port as internal and external port is not possible.

Whether you use switchable S_0 ports for internal or external communication will depend on your individual communication requirements and the existing basic connections.

Please note that the S_0 buses each require a terminating resistor of 100 ohms at each end.

Check: Is the terminating resistor switched on in the NTBA? Forum 523/524's software is used to switch on the terminating resistor. You can adjust this setting when configuring the S_0 ports in the **Configurator** on the Web console.

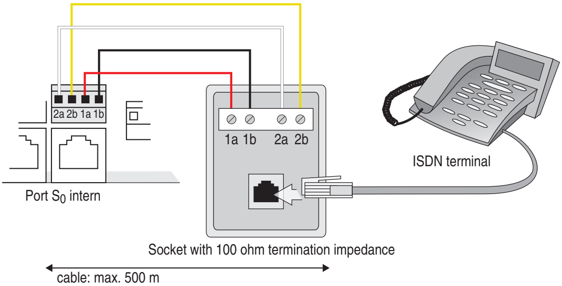
Connect $S_0$1 externally or $S_0$2 externally to the NTBA (the default setting is $S_0$1 connected to multi-terminal access and $S_0$2 connected to system access). Use the ISDN connection cable provided. For the following example, activate the **Telephony**: **Ports**: **S0(#)**: **Line termination** option in the **Configurator**. In the NTBA, 100 ohms terminating resistors between 1a and 1b as well as between 2a and 2b are required. There are no further devices connected to the NTBA.



Connecting to the network (NTBA)

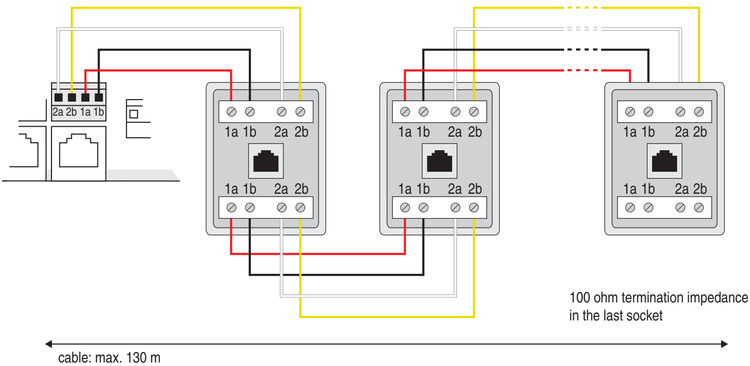
You can connect up to 8 terminals (ISDN telephones, ISDN fax machines, ISDN base stations and ISDN terminals, ISDN cards for PCs etc.) to each internal S_0 bus. The power for three of these terminals can be supplied by the bus. Where more devices are used, they will need their own power supply. The length of the four-wire cable of an internal S_0 bus must not exceed 130 m. Each internal S_0 bus has a capacity of about 3 W. The internal S_0 buses enable point-to-multi-point calls using the DSS1 protocol (Euro ISDN).

To connect an ISDN telephone, connect a pressure terminal connection to an IAE socket. For the following example, activate the **Telephony**: **Ports**: **S0 (#)**: **Line termination** option in the **Configurator**. In the IAE socket, 100 ohms terminating resistors between 1a and 1b as well as between 2a and 2b are required.

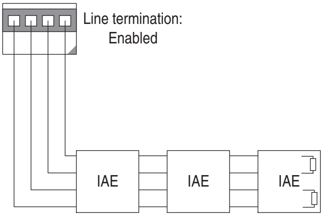


Connecting an IAE socket

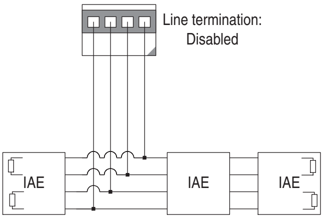
Several IAE sockets can be series connected using a bus installation. For the following example, activate the **Telephony**: **Ports**: **S0(#)**: **Line termination** option in the **Configurator**. In the last IAE socket, 100 ohms terminating resistors between 1a and 1b as well as between 2a and 2b are required.



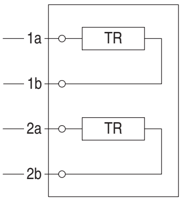
Connecting several IAE sockets



IAE = ISDN wall socket (German ISDN „Anschluss Einheit")
If the S0 bus is coming from the Forum 523/524 leading to one or more IAE sockets, the terminating resistors ("TR") in the Forum 523/524 and in the most distant IAE socket needs to be switched on.

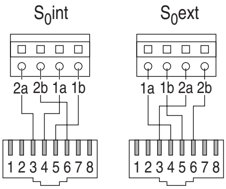


If the cabling of the S_0 bus is realized with two lines, outgoing from the Forum 523/524 and leading to several IAE sockets, the terminating resistors ("TR") in both most distant IAE sockets needs to be switched on. The terminating resistors in the Forum 523/524 must be switched off. The S_0 bus should not be installed as star topology. Sub-distributions are not allowed either.



Termination on an IAE socket: IAE sockets with built-in resistors are recommended, because the terminating resistor can be switched on or off with a mechanical switch. The figure shows how to mount resistors in an IAE socket without switchable resistors.

You can add S_0 ports by installing appropriate S_0 interface cards in a slot. These S_0 ports can be switched between internal and external operating modes.



Pressure terminals are used for internal and external S_0 ports. The connection assignment of the pressure terminals changes when switched from internal to external, as shown in the diagram above.

Tip: If, for example, you have connected an IAE socket to an internal S_0 port, and wish to switch the S_0 port to external operating mode, you will need a crossover ISDN connection cable to connect the IAE socket to an NTBA. The assignment of a crossover connection cable is described in the chapter on *PBX Networking* under the heading *Direct Connection* starting on page 156.

U_{pn} Ports

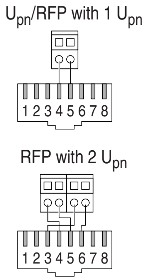
U_{pn} ports enable you to connect the digital system telephones Forum Phone 515, 525, 535 or Forum Phone 520, 530 using a twin-wire cable. You can also connect DECT base stations to a U_{pn} port.

If you want to use cordless DECT handsets (e. g. Forum Free 565 /575 or Forum Free 566 /576 /586) you will need the DECT base station Forum Base DECT. If the DECT base station is connected to a U_{pn} port, four calls can be made simultaneously using the handsets. If the base station is connected to two U_{pn} ports, eight calls can be made simultaneously. Please note however, that you can only make as many simultaneous external connections as there are externally-connected B channels available.

The length of a twin-wire cable for a U_{pn} port must not exceed a maximum of 1000 m. This cable may only be laid inside buildings.

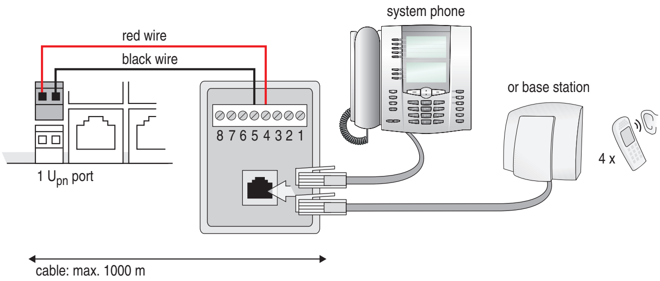
The length of a twin-wire cable for a U_{pn} port on a U_{pn} interface card may not exceed a maximum of 1000 m, where a 0.6 mm cable (with twisted pairs) is used.

Each U_{pn} port has a power consumption of about 3 W.



Pin assignment of U_{pn} ports

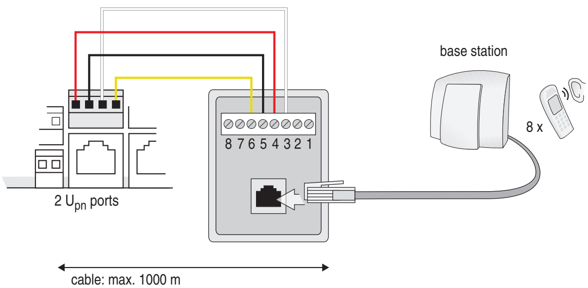
To connect a digital system telephone or DECT base station, connect the pressure terminals to a UAE socket.



If you connect a base station to a U_{pn}-port, four calls can be made simultaneously through this base station using mobile DECT devices.

Connecting a UAE socket to a U_{pn} port

A DECT base station's double connection can also be made using a UAE connection socket.



If you connect a base station to two U_{pn}-ports, eight calls can be made simultaneously through this base station using mobile DECT devices.

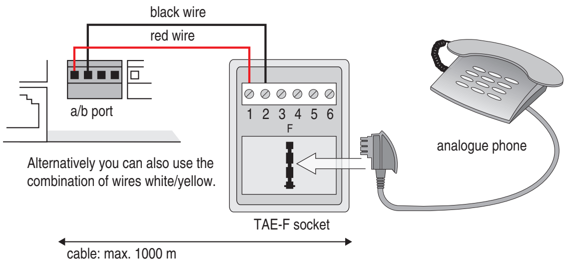
Connecting a UAE socket to a 2 U_{pn} port

a/b Ports

The a/b ports are for operating analogue devices (e. g. fax, modem or analogue telephone). The maximum permissible length of the cable is 1,000 m, where a twin-wire 0.6 mm cable (with twisted wire pairs) is used. You can then use devices for voice or data communication which use pulse dialling, or devices which use pulse or DTMF dialling, including:

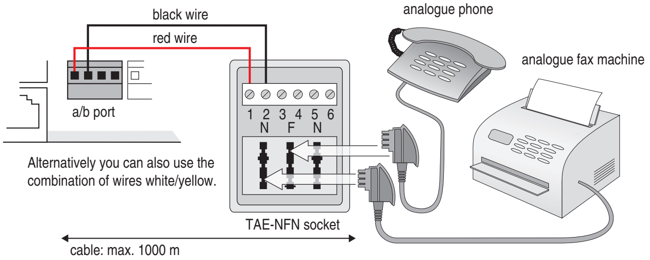
- analogue telephones,
- G3 fax machines,
- analogue modems,
- external devices for Music on Hold,
- external mail systems.

If analogue telephones are to be used, connect the pressure terminals to a standard telephone connection socket.



Connection using a telephone connection socket

To connect an analogue device (fax, modem), connect the pressure terminal to a connection socket for telecommunications devices.



Connecting a combined German standard telephone/NFN socket

Please note: *Please observe the following advice and recommendations when connecting analogue devices. Devices which do not comply with these technical prerequisites may damage the Forum 523/524!*

Analogue Telephones

If you are using analogue telephones, we recommend you use phones with DTMF dialling, because Forum 523/524's advanced features can not be used with pulse dialling

Modems

The maximum transfer rate for analogue modems is 33.6 kBit/s (V.34+).

Music on Hold

It is possible to connect a suitable external Music on Hold device. If you are not using an external MoH device, Forum 523/524 provides a preset internal MoH, for which you can also use your own melody. that you can replace in the **Configurator** of the Web console, in the **System**: **Components** menu**.** Please see the online help for further details.

Please note: *Use only devices with an input impedance of 600 ohms, and a potential-free socket for external Music on Hold. Using a device with incorrect input impedance may irreparably damage Forum 523/524!*

Mail

If you are using an external mail system, it must be able to handle the number of digits used for internal telephone numbers, e. g. five digits, if you have set up the system to use 5-digit internal numbers.

The mail system can be connected to the internal a/b ports or to the internal S_0 ports, but for both types of ports the mail system can send messages to system terminals using code procedures:
* 6 8 # [Terminal] 0 : Message exists for this subscriber
* 6 8 # [Terminal]: No message exists (without trailing “0”).

Intercom System (for a/b)

The “Doorphone” intercom system can be connected to any a/b port. The “Doorphone” module also provides the actor for the door opener contact.

Please observe the following points when connecting the door intercom system:

- The “Doorphone” module should be set to its factory setting.
- In the **Telephony**: **Ports**: **a/b**: **Change** menu in the Configurator, select **Doorstation 2-wire** as the **Type**. Deactivate the **Actor** option if you want to use the Forum 523/524 actor port instead of the “Doorphone” relay to open the door.
- The “Doorphone” intercom system has a number of bell keys to which you can assign various call numbers in the Configurator under the menu item **Telephony**: **Ports**: **Doorbell**.
- You can call the “Doorphone” intercom system by using the code procedure * 1 0 2 .
- The “Doorphone” can be connected to any a/b port, but you can only operate one “Doorphone” intercom system using Forum 523/524.

You will find more detailed information on installing and configuring the “Doorphone” intercom system in the product’s user guide.

The intercom system should only be connected by a skilled electrician, because sensor/actor contacts must be connected to the “Doorphone” module.

Actor

Forum 523/524 can be connected to a door intercom system of the “Doorphone”. This intercom system is connected to one of Forum 523/524’s a/b ports via the “Doorphone” module (see *Intercom System (for a/b)* starting on page 57).

The actor port also makes it possible to connect a separate door-opener. To do this you will need a twin-wire connection cable.

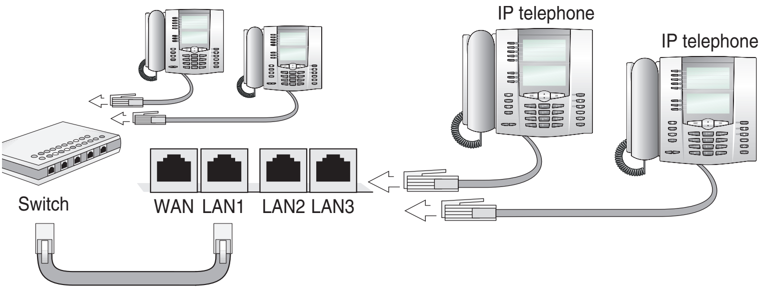
LAN Ports

The LAN ports (LAN1, LAN2 and LAN3) support 10 MBit/s and 100 MBit/s transfer rates in half duplex or full duplex operating mode. You can integrate Forum 523/524 into your company's network (LAN, Local Area Network) via the LAN ports (Ethernet ports, here provided as RJ45 sockets). This enables you to use Forum 523/524 for VoIP telephony, to operate the Web console or to use the Forum 523/524 as an IP router for connection to the Internet, among other things.

Transfer rates and operating modes are switched automatically ("Auto-Sensing function"). Connections requiring a crossover Ethernet connection cable are also automatically switched. You can therefore also use a uncrossed Ethernet connection cable to connect to a hub or switch. For VoIP application, a switch device is required.

An Ethernet connection cable (twisted pair cable in accordance with 10BaseT or 100BaseTX) may be up to 100 m long. Safe operation with 100 MBit/s transfers requires the use of category 5 cables and connecting sockets. Use a shielded Ethernet cable (STP cable, Shielded Twisted Pair cable).

There is an internal Ethernet switch with several ports on the Forum 523/524's main board. The switch is connected to LAN1, LAN2, LAN3 and slot 2. The CPU provides two separate Ethernet interfaces/ports internally. One of these interfaces/ports is connected to the LAN ports, the other to the WAN port.



Position of the LAN ports (internal switch not displayed here)

The following Ethernet ports are on the main board:

- WAN: This port is suitable only for connecting a DSL modem or connection to an external Internet Router.

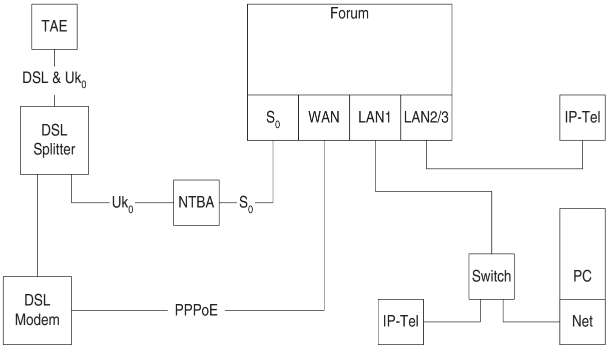
- LAN1: You should use this port for the connection to your company's network.
- LAN2 and LAN3: You should use these LAN ports to connect your VoIP system telephones.

Tip: Use one of the LAN ports, LAN1, LAN2 or LAN3 to temporarily connect a service PC.

WAN Port

Forum 523/524 can be connected to an external DSL modem and to an Internet router. The most secure Internet port, which is also separate from the LAN, is a WAN port. Transfer rates and operating modes are switched automatically („Auto-Sensing function"). Connections requiring a crossover connection cord are also automatically switched.

Connect the ethernet port of the DSL modem to the Forum 523/524's WAN port. Forum 523/524 will operate the DSL modem using the PPPoE protocol.



Forum 523/524 network connection via ISDN and DSL

Power Failure

In the event of a power failure, all configuration data, voice box messages as well as account data are preserved. The internal clock will continue to run for about 24 hours. If the power failure lasts for more than 24 hours, the time and date are reset to the factory settings when the power is switched on again. Depending on the **System**: **Common**: **Synchronize Time** setting, date and time are set to the current values on the first outgoing call as provided by the exchange.

Please note: *Do not reset the Forum 523/524 by pulling the plug. Actual configuration changes may be lost and reconnecting to a SIP provider may need a longer time. Use the code number procedure* * 1 8 5 *[System PIN]* # *instead.*

Forum Phone 515 / 525 / 535: Extensions and Accessories

Power Supply Unit

The power supply unit 90035810 is required in the following cases:

- when connecting a key extension to a Forum Phone 525 / 535 system telephone (see also the chapter *Key Extensions* starting on page 62)
- when using the Forum IPhone 525 / 535 IP system telephones (with or without key extension) where no Power over LAN is available in the network

Connecting the Power Supply Unit to an IP Telephone

The connector for the power supply unit is in the bottom of the telephone's casing and is indicated by the symbol .

1. Plug the power supply unit's RJ45 jack into the socket provided.
2. Pass the power supply unit's cable through the recesses on the underside of the IP system telephone.
3. Connect the power supply unit to the mains power supply (see *Connecting the Key Extension* starting on page 63).

Key Extensions

Up to three key extensions can be connected to system telephones: either three key extensions of the model Forum 500 Keypad Paper FP 525 / 535 or three key extensions of the model Forum 500 Keypad Display FP 535. A combination of these key extensions is, however, not possible.

The following equipment combinations are possible:

Key extension …	**… with the features**	**connectable to a system telephone**
Forum 500 Keypad Paper FP 525 / 535	– 36 keys with LED indicator – Labelling on label strips	– Forum Phone 525 – Forum IPhone 525 – Forum Phone 535 – Forum IPhone 535
Forum 500 Keypad Display FP 535	– 20 keys with LED indicator – 3 keys with LED indicator to shift levels; enables programming of 60 storage locations on each key extension – Labelling of the keys over the display; each key is assigned to a display line	– Forum Phone 535 – Forum IPhone 535

The number of key extensions connected to a system telephone (up to three) can be set in the **Configurator** of the Forum 523/524's Web Console (in the menu **Telephony**: **Ports**: **Upn** or **Telephony**: **Devices**: **VoIP Phones**). Here the keys can also be programmed as call keys or assigned functions or destination call numbers. Users can change this programming as required.

The maximum distance between the connecting socket that the telephone/key extension device combination is operating through and the Forum 523/524 must be less than 1000 metres. You will need a plug-in power supply unit no. 90035810 to provide power.

The power supply unit is plugged into the last in the series of key extensions.

Configuration	**Needs Power Supply Unit**
U_{pn} system telephone	No
U_{pn} system telephone with 1-3 key extensions	Yes
IP system telephone	Yes
IP system telephone with 1-3 key extensions	Yes
IP system telephone with PoE (Power over Ethernet)	No
IP system telephone with 1-3 key extensions and PoE	No

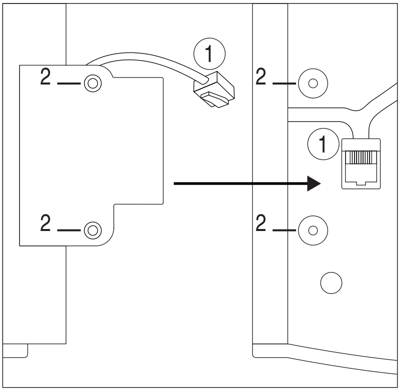
A system telephone requires a power supply unit if a key extension is installed. When using PoE, an IP system telephone requires no power supply unit.

Connecting the Key Extension

CAUTION! Guard against static charges!

Static charges can damage the Forum 523/524's electronic components. Make sure you discharge yourself and your tools before and during any installation work on the Forum 523/524 and any connected terminals. Use discharging underlays or antistatic mats where possible.

Please note: ***Never*** *attach a key extension to a system telephone that is already connected to Forum 523/524. Pull the network cable out of the socket before screwing the key extension onto it.*



Underside of the device: key extension (left) and system telephone (right)

 This symbol on the system telephone indicates the connector for the key extension. It is on the underside of the telephone. This symbol on the key extension indicates the connector for a further key extension.

 This symbol on the key extension indicates the connector for the power supply unit and is on the underside of the device. This is the same connector which can be used instead of connecting an additional key extension.

1. Plug the key extension’s RJ45 jack into the system telephone’s RJ45 socket (1).
2. Screw the key extension onto the system telephone (2).
3. Plug the power supply unit’s RJ45 jack into the socket provided on the right-hand side of the key extension.
4. Pass the power supply unit’s cable through the recesses provided on the underside of the key extension and the system telephone.
5. Connect the power supply unit to the mains power supply.
6. Connect the system telephone with the U_{pn} or ethernet port.

Headset

A headset can be connected to the Forum Phone 515 / 525 / 535 system telephones and to the Forum IPhone 525 / 535 IP system telephones.

The headset must comply with the DHSG standard (connection via RJ45 jack). The manufacturers Plantronics and GN Netcom make devices suitable for this purpose. Alternatively, you can connect a “normal” headset (RJ11 jack) using an adapter. The headset must comply with DIN Norm EN 60950-1 Point 6.2 (“Safety of information technology equipment including electrical business equipment”).

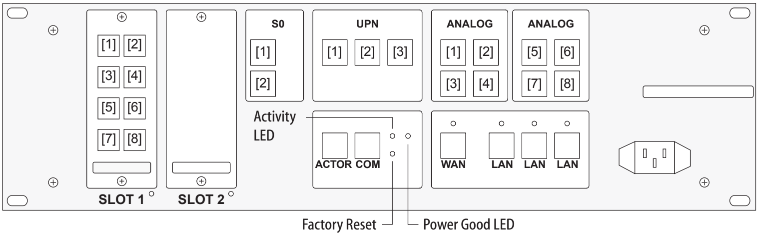
Connecting a Headset to a System Telephone

The connector for the headset is in the bottom of the system telephone’s casing and is indicated by this symbol .

1. Plug the RJ45 jack on the headset cable into the socket provided.
2. Pass the cable through the recesses provided on the underside of the system telephone.
3. Activate the headset on the system telephone in the menu **Phone settings: Headset** (see also the system telephone’s user guide).

Mounting the Forum 524 Rack InfoCom System

The Forum 524 is also available as rack version for mounting in a standard 19" EIA rackmount cabinet.



Forum 524 Rack Frame and Ports

Safety Precautions

Please note: *Before opening the device, pull out the plug.*

For the Forum 524 Rack infocom system the *Safety Precautions* starting on page 32 are generally valid. Installations, opening the housing and changing interface cards or modules is only allowed for qualified service personnel.

The system needs to be mounted in earthed cabinets or cases. Lines and cables connected to the communications system must only be laid inside buildings. Connecting a/b and U_{pn} line outside of buildings is allowed, if no internal S_0 devices are connected.

Use a shielded Ethernet cable (STP cable, Shielded Twisted Pair cable) to connect the Forum 524 Rack to a Local Area Network (LAN).

The ambient temperature of the Forum 524 Rack infocom system should not exceed 45°C. If the device is installed together with other active components, it may be necessary to mount additional ventilation fans in the installation cabinet.

Patch cables have to be connected before connecting the system to the power supply. Installation of the system, and in particular connection to the power supply and protective earthing, should only be performed by skilled, qualified personnel. EN, IEC regulations, along with other recognised technical rules regarding safety, must be observed.

Technical Data

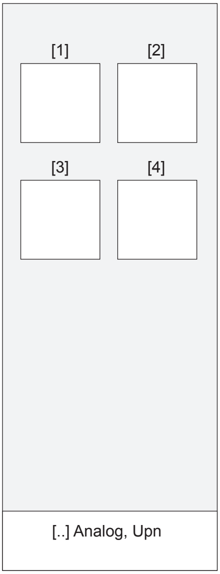
(only if different from the Forum 524)

Dimensions:	- Width: 19-inch panel with flange for mounting in installation cabinet - Height: 3 U W x D x H: 436 mm x 345 mm x 132 mm - Font panel width: 483 mm	
Weight:	approx. 8 kg	
Connections:	- Connection of 230 VAC power supply on front side via inlet connector for non-heating apparatus - Connection of all ports via RJ 45 jacks on front panel	
Actor:	Use Pin 1 and Pin 2 of the RJ45 socket.	

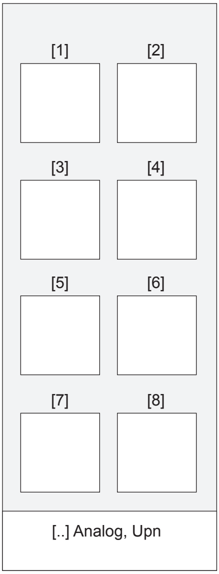
Note: *The ISDN S_0 port “S0 ext./int” on the front panel may only be switched in one direction (internal or external). Using the port as internal S_0 port requires a crossover (Rx-Tx) patch cable. Using the port as external S_0 port requires a straight patch cable.*

Mounting Interface Cards

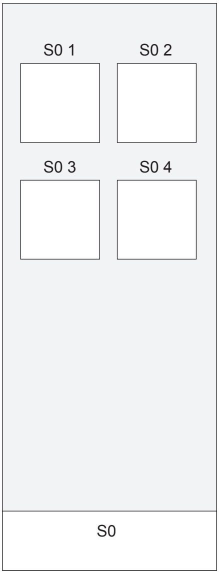
For each of the 2 slots, there is a corresponding field for mounting a metal screen with RJ 45 sockets on the front panel. The fields are labelled “SLOT 1” and “SLOT 2”. Depending on the type of interface card used in a slot, the correct metal screen has to be mounted. Besides the default screen, the following screens are available:



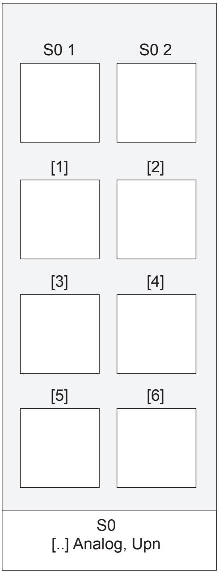
4-fold interface cards:
Forum 523 / 525 Card
4 digital extensions
Forum 523 / 525 Card
4 analogue extensions



8-fold interface cards:
Forum 523 / 525 Card
8 digital extensions
Forum 523 / 525 Card
8 analogue extensions



4-fold S_0 interface card:
Forum 523 / 525 Card
4 S0/T0



Combined interface card:
Forum 523 / 525 Card
2 S0/T0, 6 analogue extensions

An overview of the interface cards can be found under the heading *Installing Interface Cards* starting on page 40.

For each of the two slots, there is a LED on the front panel of the Forum 524 Rack. These LEDs are labelled **SLOT1** and **SLOT2**. An LED will show a constant light, if an interface card is inserted into the corresponding slot and the operating software has detected an interface card. An LED will blink, if an error condition was detected.

Scope of Delivery

- One communications system Forum 524 Rack
- One AC adapter with connection cable
- Connection cable for the ISDN S_0 port
- V.24 patch cable (connects V.24 card with front panel)
- 1 CD

Installing Interface Cards

Please note also the explanations in these sections: *Installing Interface Cards* starting on page 40. To install the card, perform the following steps:

1. Switch off the Forum 524 Rack and pull the plug out of the mains power supply. Unscrew the screws of the top cover. Take the top cover off.
2. To discharge, touch a radiator or another metallic installation connected to earth ground. Take the interface card out of its packaging. Check that it is the correct type of card. There is a label specifying the type on the connector.

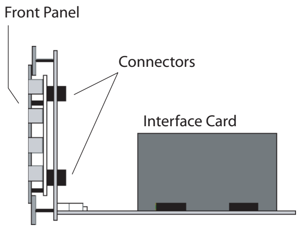
CAUTION!

Static charges can damage the electronic components of the Forum 524 Rack.

3. Carefully plug the interface card into its designated slot.

   Make sure the card is sitting firmly in the slot.

4. Install the metal screen on the front panel. Each interface card type requires a matching metal screen. An overview of available metal screens can be found under *Mounting Interface Cards* starting on page 68. Verify the tight connection of both connectors. Tighten the fastening screws.

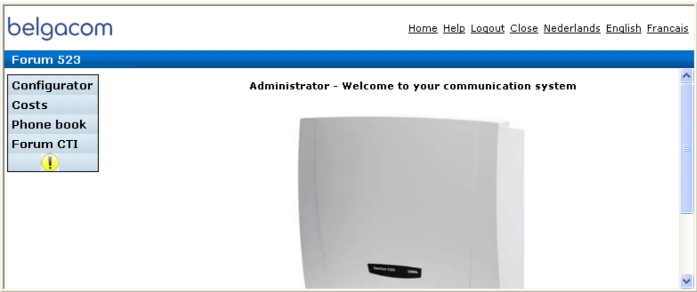


5. Close the top cover. Tighten the screws of the top cover. Switch on the Forum 524 Rack.

You can view the interface card's status in the Web console once you have restarted Forum 524 Rack. To do this, call up the menu item **Telephony: Ports: Slots**. The **inserted** column displays the interface card that is currently installed.

Configuration

Configuration and programming of the Forum 523/524 is performed by the **Configurator**, a special software application integrated into the system. The **Configurator** is operated via the Web console, which can be run on any PC connected to the Forum 523/524.



The Forum 523/524 Web console

Using the Web console, you can:

- perform the initial configuration of the Forum 523/524,
- configure users of the Forum 523/524 and authorise them to use certain system services,
- carry out further system maintenance,
- use PC-supported telephony functions,
- read out call charge information,
- access the Forum 523/524 telephone book.

The Web console has an integrated online help function that offers comprehensive information on configuration and maintenance of the Forum 523/524 (see *Loading the Online Help* starting on page 78).

For the initial configuration you can connect the PC to the Forum 523/524 via the Ethernet port. The TCP/IP network protocol is used to set up a connection via one of these ports. You can then open the Web console of the Forum 523/524 and call up the **Configurator** from there.

Note: *The Forum 523/524's IP address is always 168.99.254 in its factory settings (see LAN Factory Settings starting on page 85).*

Brief Guide to Initial Configuration

Setting up a first connection is quite simple with a standard Windows PC:

1. Connect the PC's network card with one of the Forum 523/524.'s LAN ports You can use a crossed or an uncrossed Ethernet cable to do this.

   If the DHCP automatic configuration is functioning, it is possible to continue with step 6.

2. Windows 2000/XP: log on as a user with "Administrator" rights.

3. You will find the IP settings in Windows 2000/XP under **Start**: **Settings**: **Network connections**: **Local Area Connection**. Open the dialogue box **Local Area Connection Properties**, and then the dialogue box **Internet Protocol TC/IP Properties** (see figure: *Setting the IP address in Windows XP* on page 74).

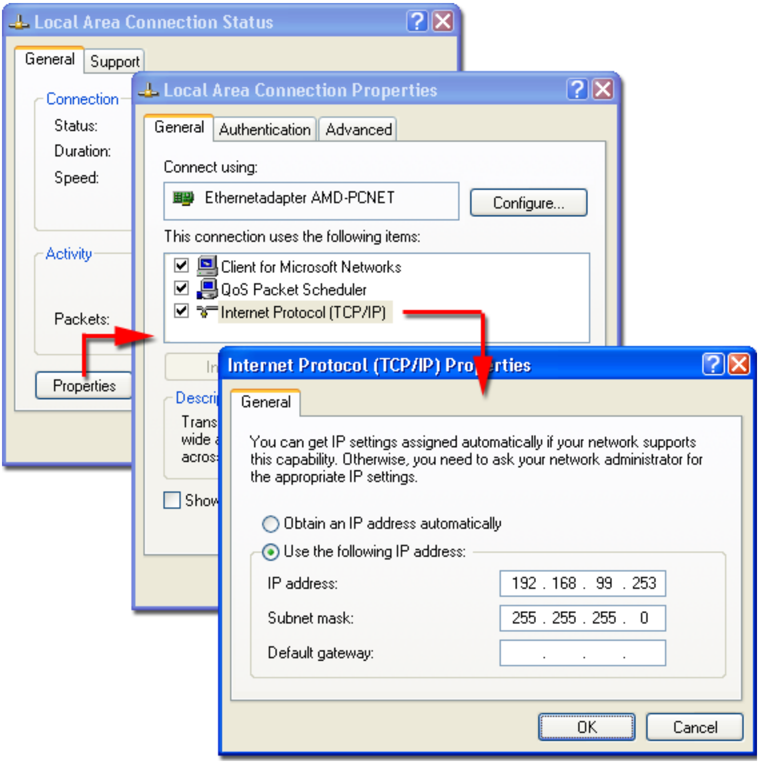
4. Note down the existing settings so that you can restore them after completing the initial configuration.

5. Change the **IP Address** to 192.168.99.253. Change the **subnet mask** to 255.255.255.0, confirm with **OK** and **Close**.

6. Start a Web browser and in the address field enter "http://192.168.99.254/".

The Web console's log-on page will be displayed. Enter the user name "Administrator" without a password for the initial configuration.

Note: *To support your next configuration steps, you should activate the **Assistant** mode on the entry page of the **Configurator**. Please also pay attention to the online help.*



Setting the IP address in Windows XP

Tip: To find out the IP address of the Web console, enter the code digit procedure ✱ 1 8 2 on a connected system telephone. You can also view the net mask by entering the procedure ✱ 1 8 3 . The PC’s IP address must be in this network range.

Note: *Deactivate any connection via a proxy server which has been configured. Open the Internet Explorer, go to the menu* ***Extras*** *and open the* ***Internet options*** *dialogue box. Select the* ***Connections*** *register and deactivate the* ***Proxy Server****.*

Configuring the Forum 523/524

Preparing the Configuration

Before starting with the configuration, make sure you have the following documents at hand:

- An overview of the ports
- A list of the terminals to be connected
- A list of the IPEIs, if you wish to log on DECT terminals in the secure procedure
- A list of the users to be set up (staff entitled to use the services of the Forum 523/524) with their names, departments, and the internal call numbers you want to allocate to them
- For Internet access: the Internet service provider access data.

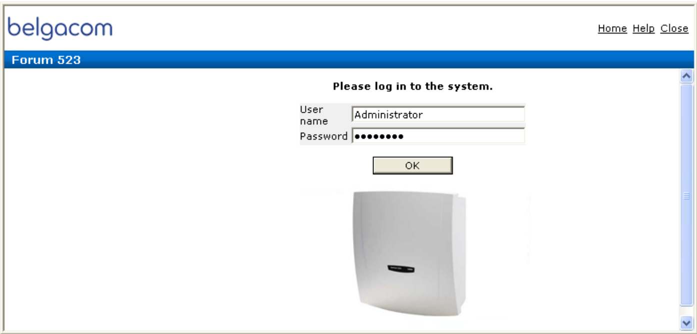
Data not available for initial configuration can be updated or corrected at a later date.

Starting the Web Console

1. Start your Web browser. Enter the Forum 523/524 IP address in the “Address” box: http://192.168.99.254/.

   If the configuration PC gets its IP address automatically from the Forum 523/524 or if the Forum 523/524 is entered as the domain name server, you can also start the Web console by entering the DNS name. The DNS name in the factory setting is **host.domain**. You can change this in the **Configurator** (**Network: LAN** menu).

2. The Forum 523/524 Web console is started. First set the country in which you are operating the Forum 523/524, and in which language the Web console is to be displayed.

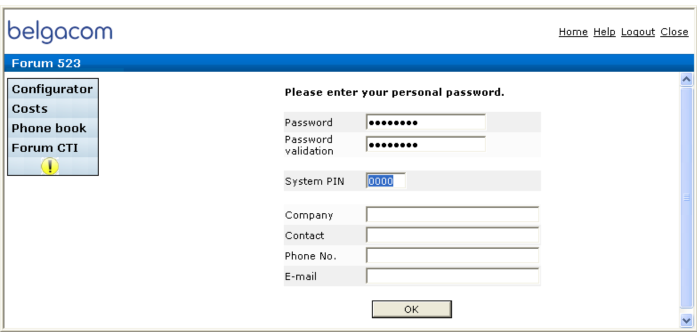


Forum 523/524: log on dialogue box

3. To commence configuration, you must first log on. For the initial configuration, enter your:

   user name: “Administrator”
   password: for the initial configuration, leave this box blank.

4. Confirm this by clicking on **OK**.



Forum 523/524: dialogue box for initial access

5. The software opens a dialogue for initial access. Determine an administrator password and enter it in this dialogue. Also fill in the other input fields.

Note: *It is imperative that you enter an administrator password to prevent other users being able to log on (without a password) as administrator. The administrator has comprehensive configura-*

tion rights, which, for security reasons, must be restricted to the administrator. The administrator password can be seen and edited only by the administrator personally. Make a note of the password and keep it in a safe place.

6. Confirm your input with **Apply**.

   Then the EULA (End User License Agreement) is displayed. Confirm the license agreement by clicking on **Accept**.

Note: *If you do not confirm the EULA, the dialogue for loading the firmware page is opened. This is where you can download an older licensed firmware version once again.*

7. Click on the **Configurator** button on the home page.

Note: *When the Forum 523/524 is commissioned, all connected terminals are assigned to the "Administrators" user group. In your initial configuration, you should configure additional user groups with restricted authorisations and assign them terminals as required by your users.*

To support your next configuration steps, you should activate the **Assistant** mode on the entry page of the **Configurator**. Please also pay attention to the online help.

Note: *Deactivate any connection via a proxy server which has been configured. Open the Internet Explorer, go to the menu **Extras** and open the **Internet options** dialogue box. Select the **Connections** register and deactivate the **Proxy Server**.*

You will find notes on using the **Configurator** and configuring **Ports**, **Users** and **User groups** in the online help. Click on **Help** in the menu bar or click on **TOC** to activate an overview of help topics.

Tip: Often a small yellow exclamation mark symbol appears below the navigation area upon the first-time commissioning. Click on this symbol to obtain information on necessary configuration steps or on the communications system status.

Loading the Online Help

The online help can now be loaded in the Configurator:

1. Go to the **System: Components** menu. Select the entry **Online Help** and click on **Browse**.
2. Look for one of the language-specific ZIP files in the “OLH” directory of the product CD. Confirm your choice by clicking on **Open**.
3. Then click on **Load** to transfer the online help to the system.

Please note: *After completion of the loading operation, the system will take a few minutes to analyse the transferred file.*

Finishing the Configuration

1. When you have completed all the settings in the **Configurator**, you must save the configuration (see also *Saving and Loading the Configuration* on page 78).
2. Then select the **Log-off** command in the upper menu bar.

Saving and Loading the Configuration

Configurations are saved in a file archive and can be loaded to the Forum 523/524 either locally from a connected configuration PC, or by remote configuration. The following configuration and customer data can be saved and loaded again:

- Telephony and network parameters
- User data
- Telephone book entries
- LCR tables

Note, that for security considerations the passwords for external SIP trunks are not re-established if you restore a data backup from another (foreign) communication system. For further information, refer to the online help documentation under the topic **System: Data backup**.

Preconfiguration

Configuration of the Forum 523/524 can be prepared by a Belgacom agent. For this purpose, a Forum 523/524 installed here is programmed with the customer data (e.g. user data, call distribution schemes, cord-bound terminals). This data is stored and then loaded into the Forum 523/524 at the customer's site by a service technician.

This prepared configuration must be completed at the customer's site (LAN configuration and DECT terminals).

For configuration of the Forum 523/524 Internet functions, first ask the responsible system administrator for details of the customer's LAN prerequisites.

Remote Configuration

The Forum 523/524 configuration can be edited or updated by the Belgacom Customer Service Centre via a remote access connection or via an Internet connection (reverse ssh tunnel). In the factory setting on delivery, two Belgacom Remote Service Centre telephone numbers are preset, under which a system technician can dial into the Forum 523/524.

Connection via ISDN

The connection for remote configuration is established using an internal RAS access. The RAS access is activated when a data call from one of the registered phone numbers is registered. During remote configuration, the Forum 523/524 is blocked for RAS access by any further users.

If no RAS access has been configured in the Forum 523/524 Configurator, you can use a code procedure to release the internal RAS access for remote configuration, and then block it again. The manual activation is automatically cancelled 30 minutes after the last configuration activity. You can add additional phone numbers for remote configuration on the configurator's **Telephony**: **Extended**: **Remote service** menu page. Activate the **ISDN** option for the desired phone numbers.

Connection via Internet

The connection for remote configuration is established indirectly over the Internet. The connection session starts with a voice call from one of the registered phone numbers. During the call, connection parameters (IP address, keys) will be sent using a DTMF sequence. Afterwards, the Forum 523/524 communications system establishes a secured SSH connection to an Internet server. The existing SSH connection allows to reverse-connect to the Forum 523/524 communications system's configurator. For this type of connection, the Forum 523/524 communications system utilizes your Internet account. Establishing an SSH connection via this Internet account must be allowed for this purpose.

The manual activation of the remote configuration activates the DTMF evaluation for the next voice call. You can add additional phone numbers for remote configuration on the configurator's Telephony: Extended: Remote service menu page. Activate the IP option for the desired phone numbers.

Note: *You also need to activate the desired type of connection directly on the* ***Telephony****:* ***Extended****:* ***Remote service****.*

Manual Activation

The following code number procedures activate or de-activate the remote configuration access. These code digits can be entered on any standard terminal or system telephone.

Remote Configuration On

* 1 9 *
(system PIN) #

Remote Configuration Off

1 9

Please note: *The system PIN is preset to "0000" and it is absolutely imperative that the system administrator changes it to prevent undesirable remote maintenance.*

In order for the system technician to be able to log into the Forum 523/524, he requires a user name and password. In this case, the solution would be to configure a "Service" user assigned to the "Administrators" user group.

Using remote configuration, all Forum 523/524 settings with the exception of the system PIN can be edited or updated. New software versions of the Forum 523/524 and the software for the connected system terminals and base stations can also be installed (see the **System: Firmware** menu in the **Configurator**).

For security reasons, settings in the **Configurator**, **Network** menu should only be edited on site to avoid malfunctions or failures in the customer’s LAN (e.g. due to IP address conflicts). Refer to the chapter entitled *Configuration Examples* starting on page 87, where interaction between the Forum 523/524 communications system and a LAN is explained.

Forced Logoff of Another User by the Administrator

If the user “Administrator” logs in and another user with administration rights is already logged in, then the administrator can forcibly log the other user off in order to configure. This functionality, for example can be used, when configuring remotely, when a user has forgotten to log out.

In order to forcibly log a user off:

1. The “Administrator” user logs-in with the administration password.
2. They open the **Configurator**.

   A message shows which user is currently configuring the system.
3. The administrator clicks on the **Take over config rights** button.

   The other user can make no further modifications of the configuration.

Codes for IP Configuration

The IP configuration of the Forum 523/524 is performed on the Web console in the **Configurator**, in the **Network: LAN** menu.

In the event that the IP configuration of the Forum 523/524 has to be changed and access via the Web console is not possible, you can also use a code digit procedure to change these basic set-

tings. Entry can be made from an analogue telephone, an ISDN telephone and from system telephones.

Set IP Address

* 1 8 2 (system PIN) *
(www) * (xxx) * (yyy) * (zzz) #

Set NET Mask

* 1 8 3 (system PIN) *
(www) * (xxx) * (yyy) * (zzz) #

Example

Enter: * 1 8 2
0 0 0 0 * 1 9 2 * 1 6 8 * 9 9 * 2 5 4 #
If required, initiate a system restart with the following procedure:

Trigger Restart

* 1 8 5 (system PIN) #

Use the PIN you entered in the dialogue box for initial access. The factory setting is “0000”.

Receiving System Messages as E-Mail

Important events and errors are kept by the Forum 523/524 in an internal log book: the error store. To inform or alert the system administrators, entries in the log book (system messages) can be sent via e-mail.

In order not to notified of every error, the administrator can define corresponding log filters (in the **Configurator**, the **Diagnosis: Log Filter** menu). These filters define which errors (category, severity, number per time interval) should be notified. The e-mails always include an internal event or error number, as well as an explanation of the message. Further, extra parameters (such as the port number when a trunk line drops out) are also provided.

The mail account for this service (**Account for LOG filter**) is configured in the **Configurator**, **Network: E-Mail** menu.

Setting up an Internet Connection from Remote (ISP Trigger Call)

(reserved tel. no.) (system PIN) * #

If the communications system is connected to the Internet via a dialup connection, a user can initiate from external that the system establishes an Internet connection (ISP Trigger call). The system is then reachable via the Internet and enables to set up a connection for a VPN connection via RAS.

Loading SW Updates

New versions of the system and terminal software can be loaded to the system.

New software versions of the Forum 523/524 are loaded from the configuration PC, which accesses the **Configurator** (see the **System: Firmware** menu). For information on connecting a configuration PC, see *Brief Guide to Initial Configuration* on page 73.

The terminal software is part of the Forum 523/524 software and is automatically loaded into the terminals via the Forum 523/524 if the software version in the terminal is different from the terminal software stored in the Forum 523/524.

For further information, refer to the online help documentation under the item **System: Firmware**.

Resetting the System Data

You can restore the factory settings of the Forum 523/524 in the Configurator. If this is not possible, refer to the next section entitled *Basic Hardware Settings Switch*.

Please note: *If this is done, all individual settings and the user data are then lost. For this reason, you should back up your configuration regularly, the best time to do so being after every change. For details, refer to the chapter entitled Saving and Loading the Configuration starting on page 78 and to the Web console online help.*

Proceed as follows:

1. In the Configurator, call up the **System**: **Restart** menu.
2. Click on **Restart with Defaults**.
3. Confirm this by pressing "OK" when the query dialogue box opens.

Basic Hardware Settings Switch

The Forum 523/524 configuration can also be returned to the factory settings by means of the basic hardware settings switch.

Please note: *If the factory settings are restored, all customer settings and user data will be lost.*

To restore the Forum 523/524 basic settings, proceed as described in the following paragraphs.

1. Switch off the Forum 523/524 by disconnecting the power plug.
2. Remove the cover (see *Opening and Closing Forum 523/524*).

CAUTION!

Static charges can damage electronic devices. Observe the regulations regarding electrostatically sensitive components.

3. The basic settings switch is designed as a key switch. The location of the switch can be found in the chapter entitled *Interface Cards* starting on page 40. Press and hold the switch.
4. Replace the power plug in the mains socket. Wait about 30 seconds until the indicator on the front of the Forum 523/524 constantly flashes.
5. Release the key switch.

   The system data is now reset.
6. Push the cover all the way down.

LAN Factory Settings

This IP address configuration is activated in the unit's factory and default settings:

Description	Setting
Forum 523/524's IP address	192.168.99.254
Forum 523/524's net mask	255.255.255.0
Forum 523/524's Host name	"host"
Forum 523/524's Domain name	"domain"
DHCP server	Active in "Dynamic and static address assignment" mode
DHCP addresses in the LAN	192.168.99.130 to 192.168.99.169
DHCP addresses via RAS/ISDN	192.168.99.10 to 192.168.99.41
DHCP addresses via RAS/PPTP	192.168.99.50 to 192.168.99.79
DHCP addresses via RAS/IPSec	192.168.99.90 to 192.168.99.119
Net mask assigned via DHCP	255.255.255.0
Gateway assigned via DHCP	192.168.99.254
DNS server assigned via DHCP	192.168.99.254
Time-Server assigned via DHCP	192.168.99.254
Domain name assigned via DHCP	"domain"

Generating Your Own MoH Files

The Forum 523/524 comes with an internal MoH file for Music on Hold. The Forum 523/524 product CD contains a number of MoH files with different volume levels, which you can load at a later time as necessary.

The file format for non-resident Music on Hold is *.wav. You can also save your own MoH in a *.wav file and load it into the Forum 523/524.

If you have a Windows operating system, you can use the “Sound Recorder” programme to generate your own MoH file. This programme is usually located in the Windows directory called “Multimedia”.

The MoH file must be coded with 8000 Hz, 8 bit mono in accordance with CCITT, A-Law. This coding is required for the Forum 523/524 and can be set in the “Sound Recorder” when you save the file under **Format** (CCITT, A-Law) and **Attributes** (8000 Hz, 8 bit mono). The maximum allowable size for a MoH file is 640 KB (approx. 80 sec. play time). If a larger file is loaded then this will be “truncated” and thereby will also only be played for 80 seconds. The MoH capacity can be subdivided in a maximum of 5 files. These files can be used for different companies or for internal and external calls.

Note: *If you don't have the Sound Recorder programme or the appropriate codec on your Windows operating system, you should install these components from your Windows CD.*

Load your MoH file in the Web console's **Configurator**, in the **System: Components** menu.

Note: *When generating your own MoH file, you may incur a fee for the use of non-resident melodies (e.g. a GEMA fee in Germany or MCPS fee in the UK). The MoH files that come with your Forum 523/524 can be used free of charge.*

Configuration Examples

Forum 523/524 in Computer Networks

One of the outstanding features of the Forum 523/524 is the integration of telephony and computer networks. Connect the Forum 523/524 via a computer network (LAN) with suitably configured workstations, and you can use its network features from these workstations. Using a Web browser you can access:

- the Forum 523/524 Configurator
- call charge administration
- the das Forum CTI, with which telephone functions can be used on a PC
- the Forum 523/524 central telephone book and your personal telephone book as well as to the company telephone book (if the multi-company variant is activated).

In addition, the Forum 523/524 can be used as an Internet access server. RAS access can also be implemented using the Forum 523/524, which enables the integration of external staff in the LAN.

In this chapter you will find several examples of configurations showing integration of the Forum 523/524 in a LAN. Which example applies to your situation depends on the size and properties of the existing or planned LAN infrastructure.

Note: *Several menu entries mentioned in this chapter are available only, if you switch on the* ***Level: Expert*** *in the top level dialogue of the* ***Configurator****.*

The following LAN prerequisites are possible:

Server configuration in the LAN	**Forum 523/524 Functions**
No IP server present	Forum 523/524 functions as DHCP and DNS server
IP server present DHCP server present	System Administrator must assign IP address and DNS name for Forum 523/524
IP server present No DHCP server present	Special case when integrating the Forum 523/524 in a LAN; settings in the **Network: LAN** menu must be coordinated with the responsible system administrator

Introduction to TCP/IP

In a single LAN it is possible to use various protocols for the transmission of data. The connection between a workstation computer and the Forum 523/524 runs via the IP protocol (also named TCP/IP) used on the Internet. IP can be used together with other protocols (e.g. NetBEUI, AppleTalk or IPX/SPX) on the same network.

Every device participating in data transmission using IP requires a unique IP address. An IP address consists of four groups of digits from 0 to 255, each separated by a full stop. The supplementary protocols DHCP and PPP automatically assign IP addresses to devices. Class C networks normally use IP addresses in which the first three numbers are the same and the last number is uniquely assigned to a specific device in the LAN. On the Internet, unique addresses assigned by a special organisation created for this pur-

pose are used. Within a LAN, you can use addresses which are not unique world-wide:

IP Range	Common Netmask	Comment
192.168.0.0-192.168.255.255	255.255.255.0	256 smaller networks
172.16.0.0-172.31.255.255	255.255.0.0	16 medium networks
10.0.0.0-10.255.255.255	255.0.0.0	1 large network

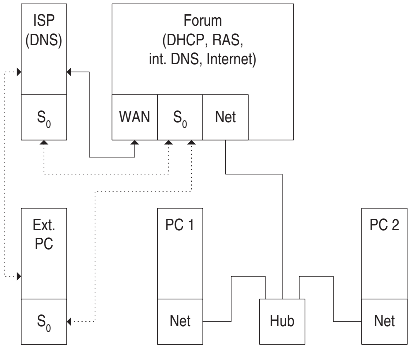
IP enables the establishment of connections via one or more intermediate stations. The decision whether to connect directly or indirectly to the partner device depends on the network mask. The network mask for a class C network is 255.255.255.0. If the IP address of the partner device does not fit the network mask, the connection is established via the default gateway. If a device knows several data routes to different intermediate stations, one speaks of a router.

The domain name system (DNS) resolves a plain text DNS name into an IP address. The DNS is a hierarchically structured database, distributed worldwide. A DNS server can supply information on the names and IP addresses for which it is responsible. For all other information, a DNS server contacts other DNS servers. For the establishment of every connection from the workstation, it is possible to give either an IP address, or a name that a DNS server resolves into an IP address.

Note: *For further explanations of technical terms, refer to the Glossary on the CD supplied.*

Forum 523/524 in a Serverless LAN

In a peer-to-peer network, the workstations are connected to one another via network cables. In many networks, the cables run in the form of a star from a central hub or switch. Such networks do not require special servers. This configuration example is also valid for a LAN with a server using a protocol other than IP (e.g. AppleTalk or IPX/SPX).



The Forum 523/524 in a serverless LAN

In a serverless LAN, the Forum 523/524 takes over the IP configuration of the connected workstations. All IP settings necessary for the workstations are assigned by the Forum 523/524 via DHCP (dynamic host configuration protocol). In this operating mode, an IP address space reserved for such networks is used:

192.168.99.254	Forum 523/524 IP address
255.255.255.0	Network mask (class C network)
192.168.99.254	DNS server IP address
192.168.99.254	Default gateway IP address

Install the IP network protocol and a Web browser for every workstation computer which is to have access to the Forum 523/524 network features.

DNS Name Resolution

In a serverless LAN, the internal DNS name resolution is performed by the Forum 523/524. If you type the string "host.domain" into your browser, a DNS request is sent to the Forum 523/524 IP address. The Forum 523/524 responds with the correct IP address, so that the **Configurator** home page can be called up.

In a peer-to-peer network (Windows network), the workstations each have a name which is displayed in the network environment. These NetBIOS names can differ from the DNS names assigned to the workstations by the Forum 523/524. The Forum 523/524 is not visible in the network environment.

Internet Access

If access to an ISP has been configured on the Forum 523/524, the Forum 523/524 can be operated as an Internet access server without any additional configuration of the workstations. When you want to see a Web page, you simply type the URL (uniform resource locator; Internet address; "http://...") in your browser. In a serverless LAN, the Forum 523/524 is configured as a DNS server and default gateway. The workstation computer therefore sends its Internet connection request to the Forum 523/524.

In almost all cases, the request will contain a DNS name which is unknown in the internal network. When you type a URL into your browser, the Forum 523/524 receives the request to find the corresponding IP address. If the name is unknown in the LAN, the request is forwarded to an ISP's external DNS server.

Note: *Workstation computers automatically add a domain name to URLs without a dot. You specify this domain name in the **Configurator**. For example, if you have configured "firm.co.uk" as the domain name, an access request for "www.firm.co.uk" will be interpreted as a local DNS request which does not lead to the establishment of an Internet connection. For this reason, you should choose a name which is not used in the Internet as the domain name ("firm-forum.co.uk" for example).*

RAS Access

You can establish a connection to the Forum 523/524 from an external PC via a VPN connection or via an ISDN card.

VPN (Virtual Private Networking) is a technology to connect an external PC over an existing Internet access using a secured and encrypted data connection. A prerequisite is the permanent internet access of the Forum 523/524. As an alternative, the “ISP trigger call” can be configured. To secure the data connection, the protocols PPTP and IPSEC are available. PPTP encryption is optional.

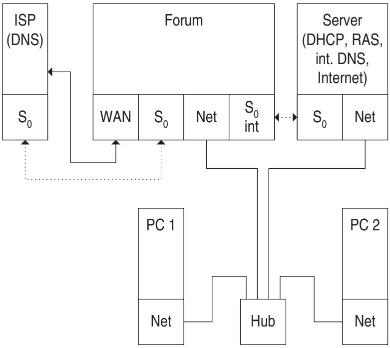
The necessary IP settings are transmitted by the Forum 523/524 on establishment of the connection. The computer that has dialled in has access to all services in the LAN that can be used via the IP protocol. The authorisation for RAS access is set up in the **Configurator** via the **User Manager: User Groups** menu.

The technical properties of the connection can be configured in the **Configurator** via the **Network**: **RAS** menu. Select one of the offered connection protocols (**ISDN**, **PPTP** or **IPSEC**). Further information can be found in the online help of the web console.

In a serverless LAN, Windows uses the NetBIOS protocol for accessing files and printers via the network environment. NetBIOS can use NetBEUI, IPX/SPX or IP as the transport protocol. In the network environment, you can only access files and printers on workstations using IP for NetBIOS.

Forum 523/524 in a LAN with an IP-enabled Server

In a LAN with an IP-enabled server, you should coordinate integration of the Forum 523/524 with the responsible network administrator. You must decide on the IP address space to be used and which network services (DHCP, DNS, RAS, Internet access) the Forum 523/524 is to handle in the LAN.



The Forum 523/524 in a LAN with an IP-enabled server

In many cases, an IP-enabled server configures the IP settings via DHCP for all workstations. In networks in which the IP settings are made manually, you have to enter the corresponding IP settings in the Forum 523/524 **Configurator** (**Network: LAN** menu). Additionally you should change the DHCP server to static address assignment (in the **Network**: **DHCP** menu) to enable the Forum 523/524, for example to configure connected VoIP system telephones. In certain cases you have to restrict the DHCP function of the IP-enabled server to ignore the MAC addresses for such terminals.

Dynamic Address Assignment for Specific Devices

In addition to the static address assignment, you can also use dynamic address assignment if you limit dynamic assignment to specific devices. You can use this e.g. to easily configure VoIP devices because you do not need to assign a fixed IP address during setup.

1. Call up the **Network**: **DHCP** page. Click **Change**.
2. In the **Status** selection, change to the **Dynamic address assignment** or to the **Dynamic and static address assignment** setting.
3. In the **Devices** selection, change to the **with configured MAC only** setting.

   For SIP system telephones (Forum IPhone 512 / Forum IPhone 545), it is also possible to omit the configuration of the MAC address (see *Forum IPhone 512 / Forum IPhone 545 DHCP* starting on page 125). In the **Devices** selection, change to the **with configured MAC only and SIP system devices** setting. Please note, that you also need to exclude from the configuration of an external DHCP server all MAC addresses starting with 00:08:5D.
4. Confirm with **Apply**.

By activating the limitation to specific devices, only known devices can get a DHCP answer. For this, the IP address is taken from the address range used for dynamic address allocation. If you setup a new VoIP device, it is sufficient to enter only the MAC address.

DNS Name Resolution

In a LAN with an IP-enabled server, the latter is also responsible for DNS name resolution. If you want to start the **Configurator** by entering a DNS name, you must link this name on the server with the IP address used by the Forum 523/524. For further information, refer to the server documentation.

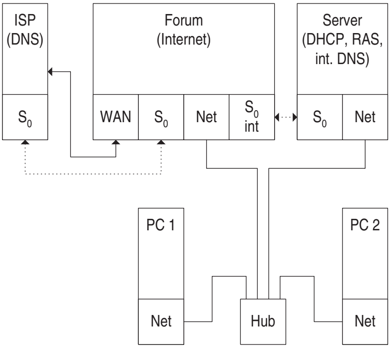
Note: *To access the Forum 523/524 under the same IP address after a restart, you must specify this IP address permanently on a DHCP server. On a DHCP server it is possible to link the MAC address of a network card with a specific IP address. You will find details in the server documentation.*

Internet Access

You can also use the Forum 523/524 as an Internet access server in a LAN with an IP-enabled server. To do this, you must enter the Forum 523/524 IP address on the server as the default gateway. In addition, you must edit the internal DNS server configuration so that the resolution of external DNS names is forwarded to the Forum 523/524.

In this example, the Internet connection is established from a workstation computer via the server, which in turn requests Internet access from the Forum 523/524.

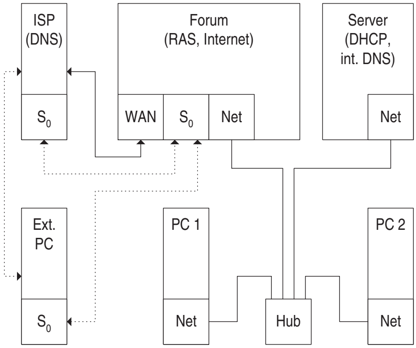
There are two different ways of suitably configuring the internal DNS server. You can enter the Forum 523/524 IP address as a DNS forwarder. If you require access to extended DNS information, you can also configure the DNS server for a recursive DNS request without the DNS forwarder. For further explanation, refer to the DNS server documentation.



The Forum 523/524 as a DNS server in a LAN with an IP server

RAS Access

In a LAN with an IP-enabled server you can also enable external computers to dial in via the Forum 523/524. To do this, you should coordinate with the network administrator the IP address space which can be assigned to an external computer dialling in, and enter it in the **Configurator**, **Network: RAS**: **PPTP/IPSEC/ ISDN** menu, under **Address Range**.



RAS access by the Forum 523/524 in a LAN with an IP server

The user account administered by the Forum 523/524, with which dialling in is permitted, only allows the establishment of direct and anonymous TCP/IP connections such as HTTP, FTP or SMTP connections. If you additionally want to allow file or printer access in the network, you must set up a suitable user account on the addressed server for network log-in. If you use the same log-in name for the Forum 523/524 user account and the same password for the network log-in, you have to enter this combination only once when dialling in.

Note: *In a larger Windows network with several segments, the lists of computer names visible in the network environment can no longer be established by broadcasts. In this case you use a special WINS server whose address the Forum 523/524 does not make known to the workstation computer when dialling in with IPSEC or ISDN. For this reason, you enter the address of a WINS server manually in the network settings of the workstation.*

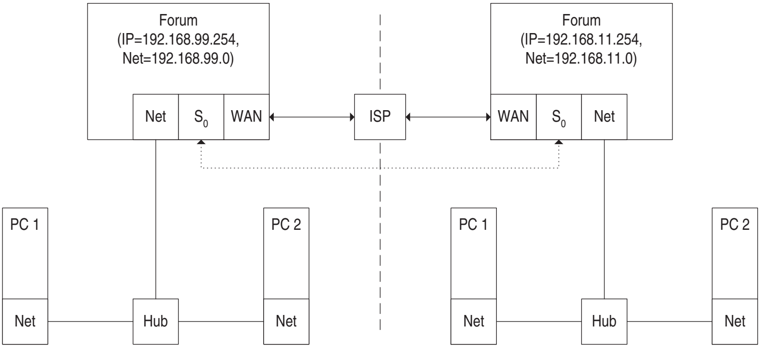
Branch Link

You can use the Forum 523/524 to interlink two LANs via ISDN or via an encrypted VPN connection (Virtual Private Network).

In case of a VPN connection, both Forum 523/524 systems use an Internet connection to transfer data. For encryption, the protocols PPTP (Point to Point Tunneling Protocol) and IPSec (Secured IP) can be used.

In case on an ISDN connection, you configure two Forum 523/524 systems so that they can dial in to each other.

In order for this to work, the two LANs must be configured for different IP address ranges (subnetworks). For at least one of the Forum 523/524 systems, change the prescribed address range for the LAN.



The Forum 523/524 in a LAN-to-LAN link

In the **Configurator**, **Network: Branch** menu you can configure the dial-in settings.

- **PPTP**: The Point-to-Point Tunneling Protocol provides a VPN setup which is easy to configure. You provide on both sides a password for the mutual authentication and encryption.
- **IPSEC**: For higher security demands, you should use this protocol to secure the VPN connection. You should transfer the keys to be entered on the configuration page via a secured channel (e.g. as disk file or via postal service).
- **ISDN**: If an IP data transfer is required to the other LAN, the Forum 523/524 initiates a dial-in via ISDN.

Note that such a connection is only set up when specific requests are made. These can be for FTP file transfers, e-mails or downloading Web pages. Name resolution via broadcasts is not possible. If you wish to use the LAN-to-LAN link to access files and printers in the Windows network, you need an IP-enabled server that administers the name resolution for the Windows network.

As the IP address range, you can select one of the 256 class C subnetworks designed for local LANs. Select a class C sub-network in the range from 192.168.0.0 to 192.168.255.0.

Useful Information on Internet Access

Costs

The Forum 523/524 uses a router function to access the Internet, which means that it automatically establishes an Internet connection when required and terminates the connection after a certain period of time if no data are being transmitted.

Unfortunately, programmes other than those typically intended to access the Internet (such as your browser or your e-mail software) may send out data packets which cause an Internet connection to be established, even if these programmes are not strictly Internet-associated applications.

Examples of such programmes are the Microsoft™ XP™ operating system, various multimedia programmes such as Realplayer™ and anti-virus applications that may establish an Internet connection for automatic updates (the so-called “phone home function”).

It is therefore highly advisable to limit ISP access by specifying the maximum monthly connection time under **Connection time per month** in the **Network**: **WAN**: **[Provider]** menu on the web console.

Using the Web

A Web browser not only enables you to use the Forum 523/524 **Configurator** from every workstation computer but also to obtain a wealth of information from the Internet. Simply enter the desired URL in the address field of the browser. Access from a stand-alone PC via an online service differs from Internet access via the Forum 523/524 in the following respects:

- When you request a Web page, dialling in results automatically. There is no display of dialogues with manual confirmation of dialling in or hanging up.
- Requesting Web pages is not a connection-orientated service. When the Web page has been loaded completely, the TCP/IP connection is cleared. If you do not request further Web pages, the Forum 523/524 automatically releases the connection to the Internet after a certain, specifiable duration.
- It is possible to call up Web pages simultaneously from several workstations.
- The Forum 523/524 can block access to certain Web pages by means of filter lists.

E-Mail

One of the most important services in the Internet is e-mail. E-mails are buffered in individual e-mail accounts on a mail server. Mail servers are operated by ISPs for example. With the Forum 523/524 you can set up one or more e-mail accounts for every user account configured on the Forum 523/524. These e-mail accounts are then checked at regular intervals.

If there are new e-mails in an e-mail account, and the Forum 523/524 has been configured for this function, the user specified in the Forum 523/524 user account is notified of the new e-mail on his system terminal. Forum Phone and Forum Free system terminals can also display information such as the sender or the subject of the e-mail.

NAT

Network address translation (NAT) is activated on accessing the Internet (ISP). You require this feature in order to translate internal IP addresses to valid external IP addresses. This has three important consequences for Internet access:

- Several workstations can share a single Internet access. You do not require a LAN access, only a single account with the Internet service provider.
- The IP addresses used in the LAN are translated into IP addresses valid worldwide. So you require no such addresses for your LAN.
- Only IP connections triggered from a workstation computer can be established. Consequently, while you can call up Web pages from a workstation, you cannot install a Web server visible in the Internet on a workstation.

Certain protocols cannot be used when NAT is being used. This affects protocols with the following properties:

- IP addresses are transported in the useful load, e.g. NetBIOS over IP or SIP.
- The protocol requires an active, inward-directed connection establishment, e.g. ICQ.
- The protocol will function without TCP/UDP port numbers, e.g. ICMP or IGMP.

The Forum 523/524 NAT has suitable processes for ensuring the functions of many important protocols affected by these rules. These are the protocols FTP (in “active” mode), CuSeeMe (“video-conferencing”), IRC (“chat”), ICMP errors (“traceroute”) and ICMP echo (“ping”).

Depending on the internet telephony protocol (VoIP, SIP) the required NAT extension (“Full Cone NAT”) or RTP-Proxy is activated on the Media Gateway Card.

Protocols which require inward-directed connection establishment can be configured in the **Network**: **Port Access** menu. For further information, refer to the online help of this menu.

Voice over IP (VoIP)

The term “Voice over IP” describes the usage of IP-based data networks for telephony. It is possible to distinguish between two different types of VoIP:

- Telephony via Internet provides cheaper charge-models for telephone services. For telephony directly via the Internet, only the cost of data transmission is incurred. Various Gateway providers can provide crossover into the PSTN (“Public Switched Telephone Network”) for a fee. As well as standardised protocols such as SIP and H.323, proprietary protocols such as the Skype network, are used. Voice and service quality via Internet is often indeterminable because they are dependent on the communication lines of various service providers. which have been optimised for data communication
- Telephony via Intranet enables joint usage of existing infrastructure for telephony and for data communication. Integrating the two communication networks into a single communications network can provide considerable savings. The Forum 523/524 gives users all the features of system telephony through its use of an IP-based protocol. Furthermore, the standardised SIP protocol can also be used on the intranet. The control of the data connections used makes it possible to define exactly the voice and service quality.

VoIP telephony over the Internet using the Forum 523/524 provides you with the following options (see also *SIP Telephony* starting on page 117):

- You can use low-cost “SIP trunk lines” with your existing Internet connection
- You can use the services of a SIP gateway service provider to access the public telephone network (PSTN)
- Automatic fallback (bundle overflow) to ISDN connections in case of the breakdown or over-occupancy of the SIP connection

VoIP Telephony via intranet with the Forum 523/524 offers the following possibilities:

- Use of IP-based system telephones and of SIP telephones connected to Cat5 twisted-pair ethernet cables
- Use of IP-based system telephones and of SIP telephones via VPN, RAS, Branch or WLAN connections
- Using voice-data compression with compressing codecs, it is also possible to make multiple IP-based telephone calls simultaneously on a 64 kbit/s ISDN line
- Use of PC-supported system telephones (so-called "Softphones") without extra hardware costs
- Operation of SIP-capable telephony software (see also *SIP Telephony* starting on page 117)
- TC system networking using Q.SIG-IP via VPN connections (see also *PBX Networking* starting on page 152)
- Setting up a DECT over IP network lets you use existing Ethernet cabling to set up a DECT network. The special DECT base stations designed for this purpose, can be handled using Forum 523/524's Web interface (see *DECT over IP®* starting on page 140).

Integrating voice and data communication within the Intranet can provide savings possible and a range of new possibilities. However joint usage of existing network infrastructure may also cause conflicts, with IP address configuration via DHCP for example (for details see *Start Procedure* starting on page 131). You should therefore always plan the use of VoIP in the Intranet together with your network administrator. In order to avoid possible conflicts please also note the information under *Fundamentals* starting on page 107.

Quick Start

IP System Telephony

VoIP system telephony can be quickly and easily set up using the Forum 523/524.

1. Optional: install a Forum 523 / 525 Media Gateway Card in slot 2 to enhance VoIP.
2. Call up the Configurator and go to the page **Telephony**: **Ports**: **Slots**. Click on the corresponding slot. Under **Configured,** select **MGC VoIP.** Optional: enter an IP address from the Forum 523/524 IP network which is not being used, such as 192.168.99.253 under **IP Address Configured**. Click on **Apply**.
3. Go to the page **Telephony**: **Devices**: **VoIP Telephone** and click on **New**. Enter the **MAC Address** printed on the underside of the IP system telephone. Select the **Type** and enter an internal **Number**. Option: enter an IP address from the Forum 523/524 IP network which is not being used. Click on **Apply**.
4. Connect the IP system telephone's LAN connection to the LAN and connect the phone to the mains power.

Once your IP system telephone has been successfully started, you can set it up and use as you would any other U_{pn}-based system telephone.

Note: *Use a shielded CAT-5 Ethernet cable (STP cable, Shielded Twisted Pair cable) to connect an IP telephone to a Local Area Network (LAN).*

External SIP Line

If your Forum 523/524 provides access to the Internet, you can an easily and quickly set up an SIP line.

1. Request at least one SIP account from an SIP provider.
2. Install a Media Gateway card (see *MGW Interface Card* starting on page 116).
3. Call up the Configurator and go to the page **Telephony**: **Ports**: **Slots**. Click on the corresponding slot. Under **Configured,** select **MGC VoIP.** Optional: enter an IP address from the Forum 523/524 IP network which is not being used, such as 192.168.99.253 under **IP Address Configured**. Click on **Apply**.
4. Call up the Configurator and go to the page **Telephony**: **Trunks**: **SIP Provider**. If your SIP provider is not listed, click on **New**. Otherwise select the preconfigured SIP provider. Enter the **Name** and **Domain** (DNS name of the SIP ID). Enter the SIP server's IP address under **Proxy/Registrar** and an IP address under **STUN Server** and **STUN Port** where necessary. You can obtain more information on this from your SIP provider. Click on **Apply**.
5. On the **Telephony**: **Trunks**: **SIP Trunks** page, click on **New**. Activate the **Status** and enter a name for the account under **Name**. Select the **SIP Provider**. Enter the relevant account information under **User name**, **Password**, **Phone No.** and **SIP ID**. Click on **Apply**.

   The **SIP ID** setting will be used while logging in to the SIP provider. Die **Phone No.** setting denotes the external phone number used within the public phone network. You can enter this number here to support system administration.
6. Call up the **Telephony**: **Trunks**: **Route**: **New** page once again. Enter "SIP", for example, under **Name**, under **Code** the number "8" and select the SIP account that was just configured for **Bundle/SIP trunks 1**. The SIP account is now available with the dialling prefix "8". To use the SIP account by default, call up the page **Telephony: Trunks: Route** and select the route **External trunk.** Under **Bundle/SIP trunks 1,** select the SIP account you have just set up. Click on **Apply**.

Note: *The **Telephony**: **Trunks** menu page is only displayed if you activate the **Level**: **Expert** option on the opening page of the **Configurator**.*

7. Check that the SIP connection is active on the **System Info**: **Telephony**: **Trunks** page. Also check the SIP licence count on the **System: Licences** page.

Check the functionality by making an external call. You should assign the relevant external number of the SIP account to the internal numbers on the page **Telephony: Call distribution: Incoming**.

Internal SIP Telephony

SIP telephones connected via LAN or SIP telephony software on LAN workstation computers can also be operated with the Forum 523/524.

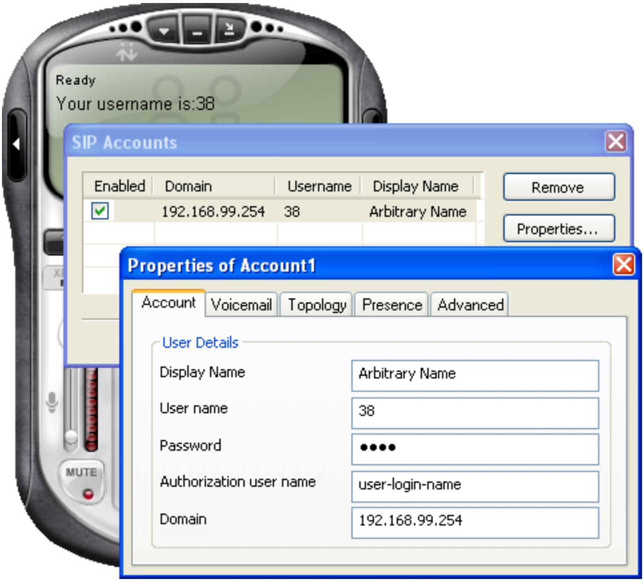
1. Install a Media Gateway card (see *MGW Interface Card* starting on page 116).
2. Call up the Configurator and go to the page **Telephony**: **Ports**: **Slots**. Click on the corresponding slot. Under **Configured,** select **MGC VoIP.** Optional: enter an IP address from the Forum 523/524 IP network which is not being used, such as 192.168.99.253 under **IP Address Configured**. Click on **Apply**.
3. Call up the **Telephony**: **Devices**: **VoIP Phones** page in the Configurator. Click on **New**. Select the “SIP” option under **Type** and enter an internal **Phone No.** Click on **Apply**.
4. Call up the **User Manager**: **User** page. Assign the new internal call number to a user.

Tip: Internal SIP telephones can also be operated by users without passwords. If you do not assign the call number of the SIP telephone to a user, you can only configure a “Guest” user account on the SIP telephone.

5. An internal SIP telephone can be operated with a dynamically assigned IP address. If the SIP telephone has its own Web interface, for example, a static IP address can be practical. Click on **New** on the **Network**: **DHCP** page. Enter the **MAC**

address of the SIP telephone and an available **IP address** and click on **Apply**.

6. Configure the SIP telephone or the SIP telephony software. Please also refer to the configuration help on the **Telephony: Devices: VoIP Phones** page. For the desired call number, click on **(Help)** and select a suitable help page under **Type**.



A configuration dialogue of SIP telephony software

7. You can only conduct a certain number of telephone calls simultaneously with internal SIP telephones. The number licenced can be viewed on the **System**: **Licences** page. The number of SIP telephones currently licenced can be determined on the **System Info**: **Telephony**: **SIP phones** page. If you click on **Reset licences**, the available licences will be reassigned with the next incoming or outgoing calls.

Fundamentals

VoIP makes the transmission of voice and telephony signalling via IP ("Internet Protocol") possible. After a connection is established, the terminal collects voice data (PCM data), which is then sent to the receiver using an IP packet. PCM data can also be compressed to save bandwidth.

Propagation Delay and Bandwidth

IP-based data networks are generally not able to guarantee a specific minimum bandwidth and defined propagation delay. A synchronised 64 kbit/s ISDN line guarantees a fixed data rate as long as the connection exists. In an IP-based data network, the data rate and propagation delay can vary. Short-term bottlenecks or retransmission due to errors may be the cause. A data flow interruption of a few seconds is barely noticeable when fetching a Web page, but it can be seriously interfere with a telephone call.

A modern Intranet normally offers enough performance reserves and reliability to make good-quality VoIP telephony possible. Specific components can also be optimised; for example by using a modern switch which evaluates the TOS byte of IP packets, by replacing unreliable connections, or by using a separated VLAN for VoIP.

Latency and Packet Length

For technical reasons, there is always a delay ("latency") between the recording of voice data via the microphone and playback via the receiver. Voice data is recorded for a short period so that it can be sent in an IP packet. The IP packet also has a signal-propagation delay before the receiver can begin playback. For these reasons, the extra time required for voice-data encoding and decoding may be neglected.

An IP packet consists of protocol data and user data. Sending shorter voice-data packets causes the ratio between the user data and the protocol data to become unfavourable and increases the bandwidth required. Sending longer voice-data packets increases latency.

The length of the voice-data packets must therefore be adjusted to the requirements of the transmission medium. Shorter voice-data packets can be sent if a direct ethernet connection exists. If an 64 kbit/s ISDN line is to be used for transmission, then longer voice-data packets should be used.

Longer voice data packages are generally used for SIP telephony over the Internet.

The following table provides an overview of the required bandwidth for a telephone connection with various parameter settings. The values apply to half-duplex ethernet; for full-duplex the values can be halved.

Required bandwidth (kbit/s) with respect to Packet Length and Codec

Packet Length (ms)	G.711 (not compressed)	G.729A approx. 8 kbit/s	G.723.1 6.3 kbit/s	G.723.1 5.3 kbit/s
20	180.8	68.8		
30		51.2	48.0	45.9
40		42.4		
50		37.12		
60		33.6	30.4	28.3
70		31.09		
80		29.2		

Note: *To ensure SIP compatibility, the older system telephones Forum IPhone 520 and Forum IPhone 530 do not support the G.723 codec any more.*

Voice Quality

The achievable voice quality depends on various factors. It is possible to optimise voice-data transmission on an existing network using the available configuration settings. Measuring the network quality may also help.

The following comparison provides a guide to voice quality with specific quality levels:

Quality Levels for Voice Transmission with VoIP

Level	Voice Comprehensibility	Comparable to
1	Very Good	ISDN
2	Good	DECT
3	Satisfactory	GSM
4	Limited	Defective GSM
> 4	Unacceptable	No Connection

When a call is set up, the terminals involved negotiate the voice-data compression ("codec") that will be used. This is the first factor that determines the achievable quality level:

- **G.711** A-Law or µ-Law (Level 1, uncompressed): The audio data of a PCM channel (64 kbit/s) is adopted one-to-one. Every VoIP terminal must support this codec. This codec can not be used with an ISDN data connection.
- **G.729A** (Level 2): Reduction to approximately 8 kbit/s.
- **G.723.1 6.3** (Level 3): Reduction to 6.3 kbit/s.
- **G.723.1 5.3** (Level 3): Reduction to 5.3 kbit/s.

Unfavourable packet length selection may reduce voice quality. The duration of the recording and not the data packet's byte count is relevant in making this selection:

- Duration <= 30 ms: optimal transmission
- Duration 40 - 60 ms: one quality-level depreciation
- Duration > 60 ms: two quality-levels depreciation

The achievable voice quality also depends on the packet propagation delay and the packet loss between the terminals involved. These parameters can be determined using the "ping" programme.

Note: *Measurements made with "ping" are round-trip propagation delays. Divide the maximum value displayed by two.*

Packet Propagation Delay and Packet Loss

Value	Quality Level	Value	Quality Level
Propagation delay < 50 ms	Optimal	Loss < 1 %	Optimal
Propagation delay 50-100 ms	0.5 level depreciation	Loss 1-2 %	0.5 level depreciation
Propagation delay 100-150 ms	1 level depreciation	Loss 2-3 %	1 level depreciation
Propagation delay 150-200 ms	2 level depreciation	Loss 3-4 %	2 level depreciation
Propagation delay 200-300 ms	3 level depreciation	Loss 4-6 %	3 level depreciation
Propagation delay > 300 ms	4 level depreciation	Loss > 6 %	4 level depreciation

Optimisation

If you detect a large fluctuation in the propagation delay during measurement, this may also cause the voice quality to deteriorate. This may indicate a defective or overloaded line caused by bit-error or collision correction resulting from retransmission by the transmission procedure.

An existing star-topology ethernet-network may uses a Hub as the central distributor of ethernet packets. A Hub repeats all ethernet packets received on all connected lines. This can cause substantial collisions and result in a high fluctuation in the propagation delay.

If this is the case, use a modern switch component. Selective forwarding of ethernet packets ("Layer 2 switching") avoids collisions. Modern switch components also evaluate the TOS byte of IP packets, thereby providing the optimal prerequisites for VoIP telephony.

Note: *The Forum 523/524 uses a TOS byte ("Type of Service") value of 0xB8 for IP packets with VoIP data. This requests "Mini-*

mise Delay" and "Maximise Throughput" for IP packets marked with this value.

Call Set-up

Various IP-based protocols are used for system telephony via the Internet protocol ("IP") (see also *Start Procedure* starting on page 131). Multiple TCP connections are made between an IP telephone and Forum 523/524 for the telephone's start procedure, registration and signalling.

All voice data are directly exchanged between IP telephones using the RTP ("Realtime Transport Protocol") protocol.

Channels on a *Media Gateway (MGW)* are allocated for making a telephone connection with an ordinary terminal or for dial tones. The MGW converts IP voice data into PCM data streams used with conventional telephony and vice versa. For this, IP voice data are exchanged between the IP telephone and the gateway.

Tip: Switching between voice data channels may cause a slight delay in some circumstances. For example: when accepting a call on an IP telephone, headset users should wait about one second before answering.

Useful Services

The type of data compression used for VoIP prevents these types of connections from using certain services. Take these notes into account especially if you want to use connections made via Q.SIG-IP or SIP:

- ISDN data services can not be used
- Faxes can only be sent using the uncompressed G.711 codec
- DTMF dial tones are only received by the other party if the uncompressed G.711 codec is used. Alternatively, DTMF dial tones can be transferred using the Internet standard RFC 2833/4733. For this, the "DTMF (RFC4733)" codec needs to be activated for the codec configuration on the **Telephony**: **Extended**: **VoIP profile** page.
- Analogue modems can not be used

Tip: Configure the actual usage for the a/b ports, e.g. set them to **Fax** or **Data (analogue)**. Connections from and to these a/b ports will then be made using uncompressed or ISDN connections where possible.

VoIP Profiles for SIP

While a call is being established and also during an ongoing connection, SIP terminals can negotiate amongst one another which codecs are to be used. SIP signalling is always being conducted via the Forum 523/524 communications system. After successfully agreeing upon a codec, the resulting RTP connection can be established between the terminals directly. The Forum 523/524 communications system can also serve as a proxy for the media data stream (“RTP Proxy function”) if a direct RTP data connection is not feasible between the terminals.

Negotiating the codecs to be used takes place as follows:

1. One of the SIP terminals sends of list of supported codecs.
2. The second SIP terminal deletes the codecs from the list which are not supported and sends the shortened list back to the first SIP terminal.

The exchanged codec lists can be filtered by the Forum 523/524 communications system, e.g. to limit bandwidth based on the VoIP profile activated for the connection or for the terminal. Please note that the negotiation regarding codecs supported also takes place for connections between SIP terminals and other terminals / subscribers. For example, the Forum 523/524 communications system uses a pre-defined codec list for a connection between an SIP terminal and a TDM port:

G.711a	G.729 (if possible)	RFC4733 DTMF

IP system phones or DECT over IP base stations participate in codec negotiation with the VoIP profile configured respectively. If no common audio codec is determined during negotiation, it will not be possible to establish the call connection.

Transparent codec interconnection

Transparent codec interconnection allows SIP terminals to use addition codecs even though the desired codec is not, or not directly supported by the Forum 523/524 communications system. Some examples are:

- G.722 audio codec for HQ audio
- H.264 video codec for transmitting images

Transparent codec interconnection is supported for internal SIP subscribers and in the context of TC system networking with SIP tie line connections.

You have to create a special VoIP profile for completely unfiltered (transparent) codec negotiation:

1. Open the **Telephony**: **Extended**: **VoIP profile** page in the configurator of the Forum 523/524 communications system.
2. Click on the **New** button to create a new VoIP profile.
3. Activate the **All codecs** option. Enter a designation into the **Profile name** input field. Use **Apply** to confirm.

Allocate this VoIP profile to the desired SIP terminal or to the desired SIP tie line. Please note that bandwidth cannot be restricted for terminals or lines with this VoIP profile. This is suitable, e.g. for SIP terminals operating on a LAN.

Codec interconnection with bandwidth restriction

You can implement bandwidth restriction with additional VoIP profiles using an extended codec list. The extended codec list contains definitions for codecs which are used by the usual SIP terminals. The codecs on this list are for usage in VoIP profiles and are not directly supported by the Forum 523/524 communications system.

You can load the extended list of codecs as a description file in the **Configurator** of the Forum 523/524 communications system on the **System**: **Components** page. The extended codec list is also a part of data back-up.

Note: *You can obtain the extended codec list as a description file-from your Belgacom agent. This file should not be edited which is why editing is not supported.*

Create a VoIP profile with extended codec entries with the following steps:

1. Open the **Telephony**: **Extended: VoIP profile** page in the **Configurator** of the Forum 523/524 communications system.
2. Click on the **Codecs** button. On the following page, select the codecs to be offered for defining a VoIP profile. Use **Apply** to confirm.

   The list contains entries with various classes (“audio”, “image”, “text”, or “video”) as well as specific bandwidth information for many entries. Particularly suitable entries are indicated as “recommended”.
3. Click on the **New** button on the **VoIP profile** page. Enter a designation into the **Profile name** input field. Determine the permissible codecs in descending order in the 10 fields. Configure the desired codecs from the extended codec list in the uppermost fields. Close the list with a fall-back selection, e.g. with “G.711 A-Law (audio)” and “DTM (RFC4733) (event)”. Use **Apply** to confirm.

Allocate this VoIP profile to the desired SIP terminal or to the desired SIP tie line. Please note that any codecs not in the VoIP profile are filtered out by the Forum 523/524 communications system during codec negotiation.

Voice Activity Detection (VAD)

You can change the codec configuration under **Telephony**: **Extended**: **VoIP profile**. You can also activate the **Voice Activity Detection** option for one or more codecs in so doing. If this option is active then pauses in the conversation will be replaced with empty packets. This can be better compressed by compression-codecs thus reducing the required bandwidth. This option must be deactivated if you want to assign the profile to a SIP trunk and if you have set the G.711 codec.

Note: *If you want to allocate the VoIP profile to a Forum IPhone 512 / Forum IPhone 545 IP system telephone, the* ***Voice Activity Detection*** *option should be deactivated for all codecs.*

Media Gateway (MGW)

The Media Gateway transforms VoIP voice data into PCM audio data. This function converts voice data between VoIP telephones and all other terminal types. Without the Media Gateway, VoIP telephones can only exchange call data directly with other VoIP telephones. Media Gateway functionality is also required for producing dial tones and making external phone calls with a VoIP telephone.

A Media gateway card makes 8 channels available. One Media gateway channel should be available for a maximum of 3 VoIP terminals. The Media Gateway also takes over the routing function for external SIP connections, making 8 external SIP connections possible.

Software MGW

The system software for Forum 523/524 provides a Media Gateway function. Depending on the system processor utilisation and available system memory up to 32 MGW additional channels may be available.

The Media Gateway function provided by the system software has the following features:

- Up to 32 channels can be used simultaneously.
- The MGW channels are not compressed, so only the G.711 codec is available.
- There is no echo suppression.
- Voice quality may be reduced during high system utilisation.
- Q.SIG-IP, DECT over IP and SIP are not possible.

For optimal call quality and high availability, you should consider using a MGW interface card (see also *MGW Interface Card* starting on page 116).

The Media Gateway function (“Software MGW”) is used preferably for connections with the G.711 codec - even if a MGW interface card is installed.

MGW Interface Card

A Media Gateway interface card providing 8 simultaneously-usable Media Gateway channels is available for Forum 523/524.

Technical Data

- The MGW interface card is connected to the internal ethernet switch via the slot. No external ports operating via pressure terminals are provided.
- A MGW interface card can be installed in slot 2 of Forum 523/ 524.
- The MGW interface card supports all the codecs, as well as the silence detection, echo suppression and DTMF tone detection used by VoIP telephones.
- The MGW interface card has the required software stored in its Flash memory. The software is updated automatically with a system software.

Information on Use

The MGW interface card must be correctly inserted and configured (see also *Interface Cards* starting on page 40).

Each MGW interface card requires its own IP address. This can either be statically assigned or obtained via DHCP.

1. Call up the **Configurator** and open the page **Telephony**: **Ports**: **Slots**.
2. Click on the slot number in the column of the table containing the desired interface card.
3. Enter the desired static IP address in **IP Address configured** field. Enter “0.0.0.0”, to obtain an IP address via DHCP.
4. Confirm the setting with **Apply**.

The MGW interface card’s MAC address is displayed on the configuration page. You will need this for static IP address assignment by the DHCP server.

SIP Telephony

The SIP Internet (Session Initiation Protocol) protocol provides you with a low-cost, standardised option for telephoning via IP-based networks. The Forum 523/524 enables you to use external SIP telephone connections ("SIP trunk lines"). The Forum 523/524 communications system supports configuration of standard SIP lines with one call number for each as well as extension-capable SIP-DDI lines with base call number and extension.

Furthermore, internal SIP subscribers, SIP telephones or SIP telephony software are also supported (see also *Quick Start*: *External SIP Line* starting on page 104 and *Internal SIP Telephony* starting on page 105).

External SIP Connections

The **Telephony**: **Trunks**: **Route** menu gives you the option to configure a bundle overflow, which automatically occupies a second line in case of a breakdown or over-occupancy of the SIP connection. You can also set up your system to route certain types of calls, such as international calls, to an SIP connection.

Note: *You will need a Media Gateway card for SIP telephony.*

You will also need a fast Internet connection such as DSL for SIP telephony.

You will also usually need the services of a SIP provider. A SIP provider operates a special server (the SIP Registrar) to handle connections. The SIP provider also operates a gateway to the ordinary telephone network which users pay to use and which enables the SIP provider to provide calls to the telephone network. A SIP connection can also accept incoming calls from the telephone network.

The same voice transmission techniques as those explained in *Fundamentals* starting on page 107 are used for SIP telephony. SIP telephony has the following distinctive features:

- Subscribers are identified through an e-mail-like "SIP ID" such as 12345@domain.net or name@sip-provider.com.

- SIP transmits dialling numbers always in a single data package (block dialling). Dialling can therefore be concluded with the hash key # on the system terminal, or the end of the number will be indicated by a time-out. The value for this time-out can be defined for each SIP provider separately.

Tip: The Forum 523/524 communications system can have a clip-board for keeping track of the most recently dialled call numbers, optimising en-bloc dialling. To do so, activate the **Dial out cache** option for a SIP line or a SIP tie-line bundle.

- You must log on ("Login") to the SIP registrar before you can use SIP telephony. Use the Forum 523/524 to manage important information for the registration (user name and password) of one or more SIP accounts. It is possible to make several calls simultaneously using a single SIP account.
- A SIP connection causes constant Internet data traffic, so do not use SIP with Internet access which is paid for according to the time used.
- RTP call data is also exchanged directly between terminals for SIP telephony, so different codecs can be used for sending and for receiving. It is also possible to change codecs dynamically during a call. You should use every codec available in the VoIP profile at least once, because this will enable you to establish connections with as many SIP subscribers as possible.
- Fairly large packet lengths are quite normal on the Internet. They compensate for the longer packet propagation delay.
- A bidirectional RTP data stream with a dynamically-assigned UDP port number is used to set up calls between subscribers. For this reason, incoming RTP calls often fail to get past the Firewall or NAT configuration of the Internet gateway product used. If you do not use the Forum 523/524 as the Internet gateway, the product should be compatible with SIP telephony. These products provide a "Full Cone NAT" setting for this application.
- To enable the use of multiple devices on a single Internet connection, the IP addresses used in a LAN (often 192.168.x.x) are translated to a valid IP address using address translation (NAT - Network Address Translation), but no status information is available for NAT on an incoming RTP

connection. To avoid this problem, the IP address of a workstation computer or telephone visible on the Internet is determined using a STUN server (STUN: Simple Traversal of UDP over NAT). You can ask your SIP provider for the STUN server's IP address and port number. If you don't need a STUN server, leave the **SIP Provider** field empty.

- For direct SIP telephony using Forum 523/524, only SIP IDs consisting of numbers for identifying subscribers registered with the SIP provider specified can be addressed
- You can integrate an external SIP connection in the **Telephony**: **Trunks**: **Route** menu into the route configuration. You can use a network provider rule to specify the routing of numbers within a specific range to use SIP telephony as a preference (see also *PBX Networking*, under *Configuration* starting on page 161).

You can configure SIP connections in the **Configurator** on the pages **Telephony**: **Trunks**: **SIP provider** and **Telephony**: **Trunks**: **SIP trunks**. Enter the technical attributes of a specific SIP provider, such as the IP addresses for the registrar and the STUN server under **SIP provider**. Under **SIP trunks** enter the information for an existing SIP account, such as the user name, password, assigned call number and the maximum number of simultaneous calls possible.

For extension-capable SIP-DDI lines, you can, as needed, make additional settings on the **Telephony**: **Trunks**: **SIP Provider** page. Assigning extension call numbers and internal subscribers is configured on the pages under **Telephony**: **Call Distribution** (**Incoming DDI** or **Outgoing DDI**). Please note the explanations on these pages in the online help. Furthermore, assigning extensions using placeholders ("wildcards") is possible. The online help offers an explanation under the keyword **Call Number Mapping**.

Internal SIP Subscribers

The Forum 523/524 becomes available as the SIP server for internal SIP subscriber telephony switching services. SIP telephones connected via LAN or SIP programmes installed on workstation computers can thus establish connections to all other devices or

trunks connected to the Forum 523/524. For operation as a SIP server a *MGW Interface Card* is required.

Licence Assignment

The number of possible SIP subscribers is determined by the number of licences purchased. In order to provide you with the greatest possible flexibility regarding usage of available licences, licence assignment is dynamic via the “Floating licence”. Using a user/password combination (“SIP log on”) you can have several SIP subscribers under the same call number. Only every new SIP log-on occupies a new licence. The technical log-on process of a SIP subscriber with a valid user name and correct password is always successful. Only when a connection is established is there an attempt made to occupy a licence under the SIP log-on. If all licences are occupied currently, the SIP subscriber can only make emergency calls.

Note: *If the technical log-on is not successful due to an incorrect user name or incorrect password, the SIP subscriber cannot establish any connections – no emergency calls either.*

When a SIP subscriber logs off, when terminating the programme, for example, the associated licence will become available immediately. A licence is also made available when a SIP subscriber’s regular status query is not conducted. The internal clock for automatic log-out is determined by the **Profile** assigned under **Telephony**: **Devices**: **VoIP Phones**. The (**Keepalive**) internal clock setting is located on the **Telephony**: **Extended**: **VoIP profile** page.

Detailed information on the current licence assignment and on logged-on SIP subscribers is located on the **System Info**: **Telephony**: **SIP phones** page. This page is where you can reset licence assignment at any time by clicking on **Reset licences**.

Technical Notes

The names of settings for the various SIP telephones or SIP programmes are not uniform unfortunately. Please refer to the **(Help)** on the **Telephony**: **Devices**: **VoIP Phones** page and the following notes when configuring SIP subscribers:

- The “REGISTER” SIP message must be sent to the IP address of the Forum 523/524 using the 5060 destination port. For SIP

subscribers, this setting is frequently located under “SIP Server” or “SIP Settings” with the terms “Domain”, “Server IP” and “Server Port”.

- The “REGISTER” SIP message must contain a valid user name and the appropriate password (the **User name** and **Password** fields in the **Configurator** under **User Manager**: **User**). For SIP subscribers, this setting is frequently located under “SIP User Settings” or “SIP Account” with the terms “Authorization User” and “Password”.
- The “REGISTER” SIP message also contains a SIP-URI in the spelling for e-mail addresses, for example `"Displayname" <sip:123@192.168.99.254>`. The text portion of the SIP-URI (“Display Name”) is not evaluated at log-on from the Forum 523/524. The series of characters before “@” is the “User Name” or “SIP Username”. The internal call number of the user must always be used here (the **Ph.No.** field in the **Configurator** under **User Manager**: **User**). The series of characters after “@” is the “Domain Name” or the “SIP Domain”. The IP address of the Forum 523/524 must always be used here.
- A STUN server (Simple Traversal of UDP over NAT) or a SIP proxy is not required because internal SIP subscribers on the LAN are directly connected to the Forum 523/524. Switch these functions off if possible.
- With a SIP terminal, you can enter an international phone number with a leading plus. When you enter a call number in the E.123 format, the plus char is substituted by the “00” number sequence and the immediate line seizure via the standard route is activated for the call. If you prefer to dial in this number format, you should activate the international call number conversion (see *E.164 conversion* starting on page 165).

Features

SIP subscribers can establish connections to all other terminals and trunks. The SIP protocol generally works with block dialling. This is why the selected call number is only activated after an internal clock has expired or activated immediately via the hash key (“#”) when dialling. This is why code number procedures

without the hash key and code number procedures with a concluding hash key can be used. An overview of code number procedures that can be used is located in the Configurator on the **System Info**: **Codes** page**.** Activate “SIP phones”. Please also note the corresponding information in the “Forum 500 / Forum 5000 – Standard Terminals” user guide.

Alongside code number procedures, SIP subscribers can also use a series of functional features realized via the SIP protocol. The Forum 523/524 is the ending for all SIP connections as opposed to what is usually the case on the Internet. This enables SIP subscribers to use Forum 523/524 features. Direct data exchange is thus not possible between two SIP subscribers. The following table shows the possible features.

Features	**Notes**
Incoming and outgoing calls with call number display (CLIP)	A SIP telephone requires a call number display for CLIP.
Parallel connection of multiple SIP subscribers	SIP subscribers must be logged on under the same user identification.
Enquiry, toggling, call waiting, three-way conference, reject	Operation or feature must be available on the SIP telephone or in the SIP software.
Call transfer	before and during a call; operation must be available.
Blind Transfer	SIP only: forwarding an incoming call without accepting the call; feature must be supported by SIP telephone or by the SIP software.
Keypad as “INFO” message	DTMF tones cannot be securely transferred “in band” via compression codecs. Digital “out band” transferral as “INFO” SIP message or via RFC 2833/4733 is supported. This feature must be available and activated on a SIP telephone or in the SIP software.

Forum IPhone 512 / Forum IPhone 545 SIP Telephones

You can operate the Forum IPhone 512 and Forum IPhone 545 SIP system telephones on the Forum 523/524 communications system. The firmware of the communications system already includes the matching firmware files for the following SIP telephones:

Forum IPhone 512 and Forum IPhone 545 SIP telephones

Model	Short description
Forum IPhone 512	Basic level version, display with 3 lines, 8 programmable keys, 2 line keys, 802.3af (PoE) power supply, two Ethernet ports for PC and LAN, separate 48 volts wall power supply available
Forum IPhone 545	Premium version with graphic touchscreen display, 802.3af (PoE) power supply, two Gigabit-Ethernet ports for PC and LAN, 11 softkeys on 5 levels, 3 line keys, up to 3 key extensions with function keys or 3 key extensions with softkeys

The Forum IPhone 512 / Forum IPhone 545 SIP telephones provide, besides the VoIP telephony, additional system telephony features which you can configure conveniently and safely with the help of the Web configurator. The Forum IPhone 512 / Forum IPhone 545 SIP telephones support G.711 codec (a-Law and µ-Law), the G.729 codec and the transmission of DTMF signals according to RFC 2833.

An *MGW Interface Card* is required to operate Forum IPhone 512 / Forum IPhone 545 SIP telephones with the Forum 523/524 communications system.

The general commissioning takes place in the **Configurator** of the Forum 523/524 communications system with the following steps:

1. Add a new entry under **Telephony**: **Devices**: **VoIP Phones**. Enter a call number and select the telephone type as Forum IPhone 512 / Forum IPhone 545. Optionally enter the MAC address and the IP address of the Forum IPhone 512 / Forum IPhone 545 (see *Forum IPhone 512 / Forum IPhone 545 DHCP* starting on page 125).

Note: *If you want to allocate the VoIP profile to an Forum IPhone 512 or Forum IPhone 545 SIP system telephone, the* ***Voice Activity Detection*** *option should be deactivated for all codecs.*

2. Under **Telephony**: **Devices**: **System phones** configure the common settings, the programmable keys and the softkeys for the Forum IPhone 512 / Forum IPhone 545 (see *Forum IPhone 512 / Forum IPhone 545 Setup* starting on page 124).
3. Connect the Forum IPhone 512 / Forum IPhone 545 to your network and plug in the power supply. Details about this can be found in the installation manual which is provided with the Forum IPhone 512 / Forum IPhone 545.

If the Forum IPhone 512 / Forum IPhone 545 has the factory default configuration, it will request an IP address configuration from the DHCP server of the Forum 523/524 communications system. The IP address and the location for loading files from the TFTP server of the Forum 523/524 is received as a part of the DHCP answer.

Now the Forum IPhone 512 / Forum IPhone 545 reads in the configuration files from the TFTP server. This includes a generic configuration file and a device specific configuration file for the given MAC address. During this process the Forum 523/524 communications system transfers all settings and by this it configures the system telephony features for the Forum IPhone 512 / Forum IPhone 545. If necessary, The firmware stored in the Forum IPhone 512 / Forum IPhone 545 as well as the diverse language modules are updated also.

Note: *The configuration files are sent using an encrypted format (file extension: *.tuz).*

Forum IPhone 512 / Forum IPhone 545 Setup

You can change the configuration and function key settings for each Forum IPhone 512 / Forum IPhone 545 individually:

1. Call up the **Telephony**: **Devices**: **System phones** menu page in the **Configurator**.

2. In the **Devices** list, select the desired Forum IPhone 512 / Forum IPhone 545.

   The menu page shows the current configuration, the function key assignment and a device graphic.

3. Click on the **Change** button to determine general settings, e.g. the display language. Confirm the setting on the following page with **Apply**.

4. Depending on the terminal type, different lists with function keys will be shown:

   – **Programkeys**: You can label this function keys on the device using a paper strip.

   Click on the function key heading to call up the configuration dialogue for this function key. Select a function in the **Type** setting and optionally enter a label in the **Labelling** setting. Confirm the setting with **Apply**.

5. To transfer the settings to the Forum IPhone 512 / Forum IPhone 545, click the **Apply** button on the **Telephony**: **Devices**: **System phone** page.

   The Forum IPhone 512 / Forum IPhone 545 restarts and thereby takes over the new configuration.

Details on using the Forum IPhone 512 / Forum IPhone 545 SIP telephones as well as an overview about the possible function key assignments can be found in the “Forum IPhone 512” and “Forum IPhone 545” user guide.

Forum IPhone 512 / Forum IPhone 545 DHCP

The address and device configuration is transferred from the Forum 523/524 communications system to the Forum IPhone 512 / Forum IPhone 545 SIP phone using the DHCP protocol. You can configure the necessary address settings while adding a device entry under **Telephony**: **Devices**: **VoIP Phones** differently:

- You can enter the MAC address of the Forum IPhone 512 / Forum IPhone 545 manually. Depending on the operation mode of the DHCP server, you can also enter the IP address manually. For this, change the **Status** setting on the

Network: **DHCP** page to “Static address assignment”. Optionally, you can use the dynamic IP address assignment from the DHCP server. For this, change the **Status** setting on the **Network**: **DHCP** page to “Dynamic address assignment” and also change the **Devices** selection to the “All” setting or to the “with configured MAC only” setting.

- By using the “Easy Configuration / Hot-Desking” feature you can add a device entry without entering a MAC address and without entering an IP address. An unregistered Forum IPhone 512 / Forum IPhone 545 will show a login page after startup. A user can enter the desired call number and the matching user PIN which will establish the association of the user account to the device entry. For this, change the **Status** setting on the **Network**: **DHCP** page to “Dynamic address assignment” and also change the **Devices** selection to the “All” setting or to the “with configured MAC only and all SIP system devices” setting.

If you want to use both features concurrently, change the **Status** setting on the **Network**: **DHCP** page to “Dynamic and static address assignment”.

If you operate another DHCP server in your network, the Forum IPhone 512 / Forum IPhone 545 can distinguish the different DHCP answers and will normally take its configuration data from the Forum 523/524 communications system nevertheless. In a conflict situation, you may need to prevent the configuration of the Forum IPhone 512 / Forum IPhone 545 SIP phone by a foreign DHCP server. For example, add an exception rule for all MAC addresses starting with 00:08:5D.

The TFTP server of the Forum 523/524 communications system will serve up to 60 concurrent data transfers. To avoid overloading the TFTP server (e.g. after a power failure), additional devices may get an IP address from the DHCP server with a delay. The allocation of IP addresses will restart if the TFTP server indicates the end of the overload situation.

Forum IPhone 512 / Forum IPhone 545 Hot Desking

The Forum IPhone 512 / Forum IPhone 545 SIP phones support the “Easy Configuration / Hot desking” feature. This feature allows a user to operate his personal configuration on any desired Forum IPhone 512 / Forum IPhone 545 of equal type.

To prepare a Forum IPhone 512 / Forum IPhone 545 for Hot Desking, you can configure a function key with the “Logout” function. If this function key is pressed, the currently assigned device entry will be marked as “Logged Out” and the Forum IPhone 512 / Forum IPhone 545 restarts. You can indicate logged out device entries in the Configurator on the **Telephony**: **Devices**: **VoIP Phone** page by their shortened MAC address.

After restart, the logged out Forum IPhone 512 / Forum IPhone 545 will show a login page. A user can enter his call number and his user PIN. If a device entry with this call number exists and has the same device type, the login procedure continues. If the device entry is currently active on another device, the associated remote Forum IPhone 512 / Forum IPhone 545 will be logged out automatically and will show a login page on its part. The now logged in Forum IPhone 512 / Forum IPhone 545 starts and will get its configuration from the newly assigned device entry.

Note: *When adding the device entry, it is also possible to activate the **Logged out** option.*

VoIP System Telephones

The following telephones and software packages are available for VoIP system telephony:

- Forum IPhone 525: This is a VoIP-enabled edition of the Forum Phone 525 system telephone. This system telephone can be extended with up to three key extensions (Forum 500 Keypad Paper FP 525 / 535).
- Forum IPhone 535: This is a VoIP-enabled edition of the Forum IPhone 535 system telephone. This system telephone can be extended with up to three key extensions (Forum 500 Keypad Paper FP 525 / 535 or Forum 500 Keypad Display FP 535).
- Forum IPhone PC: This VoIP software offers the functionality of an system telephone using Windows executable software (see *Forum IPhone PC* starting on page 137). This software also provides local answering machine functionality and can be integrated into CTI applications.
- The older VoIP system telephones Forum IPhone 520 and Forum IPhone 530 can still be used. With the Forum IPhone 520 and Forum IPhone 530 telephones no postdialling via DTMF is supported for ISDN or SIP lines.

Device Properties

The VoIP-enabled versions of the system telephones Forum IPhone 525 and Forum IPhone 535 offer the same features as the corresponding system telephones. Using VoIP system telephones is therefore not much different from using standard system telephones. The following differences exist:

- Two RJ45 connector ports are available for ethernet connection. The ports are connected to one another via the telephone’s internal switch. The switch supports 10 Mbit/s or 100 Mbit/s full-duplex with priority given to VoIP data transmission.

  LAN Port: Allows the telephone to connected to the LAN. Use a non cross-over RJ45 patch cable to connect to a Hub or Switch.

PC Port: Allows the telephone to be connected to a workstation computer. Use a non cross-over RJ45 patch cable to connect to the PC's network port.

- The VoIP system telephone's power supply is provided by an extra plug-in power supply. It is also possible to provide a power feed via PoE ("Power over Ethernet"). PoE requires special devices for power feeds, as well as a completely wired RJ45 connection line.
- You can also connect a standard headset via RJ45 sockets (DHSG standard) to VoIP system telephones.
- VoIP system telephone's audio signals are generated by the telephone itself. DTMF dial tones and Music on Hold are produced by the Media Gateway function.
- A VoIP system telephone can also be operated without a permanent connection to the communications system, for example via an on-demand RAS connection.
- Signalling data for call control, call data during three-way conferences, connections to conventional terminals and external connections is exchanged between the VoIP system telephone and the communications system. During a call between two VoIP system telephones, call data is exchanged directly between the two VoIP system telephones.
- During the device's start procedure, the IP address is configured and the device software is requested via the DHCP and TFTP network protocols.

VoIP System Telephone Configuration

The VoIP system telephones Forum IPhone 525 and Forum IPhone 535 obtain the required IP address configuration and operating software via the DHCP, BOOTP and TFTP IP protocols. After the power supply is assured, the device's internal boot loader is started which controls the further start procedure.

Standard operating procedure is to contact the Forum 523/524's DHCP server so that the start procedure can be concluded without problems. To register a new VoIP system telephone, proceed as follows:

1. Temporarily remove the VoIP system telephone’s ethernet connection. Switch on the VoIP system telephone’s power supply. Note the MAC address shown in the display, for instance “MAC: 00:30:42:00:00:00”. Switch off the power supply.
2. In the **Configurator**, open the **Telephony: Devices: VoIP Phones** page. Click on the **New** button.
3. Select the VoIP system telephone’s **Type** and enter the previously noted MAC address. Assign a **Name** and **Phone No.** Confirm with **Apply**.
4. Connect the ethernet connection with the VoIP system telephone’s RJ45 connector. Switch on the power supply. Verify the correct start procedure on the display.

LAN DHCP Server

If the LAN already uses a DHCP server to configure workstation computers, there are various options for correctly responding to VoIP system telephones’ DHCP, BOOTP and TFTP requests. A comparatively simple approach is described here.

1. Configure the LAN’s DHCP server to ignore DHCP requests from the VoIP system telephones. With a Linux DHCP server programme, you must, for example, include the following lines in the system file “/etc/dhcpd.conf”:

   ```
   group {
     deny booting;
     host 192.168.11.12 {
       hardware ethernet       00:30:42:00:11:22;
     }
   }
   ```

   Every DHCP service programme has similar options. You may need to reserve a free IP address for each VoIP system telephone. You will find more details in your DHCP service programme’s online help or handbook. The MAC address of all VoIP system telephones always begins with 00:30:42.

2. Configure a fixed IP address for the Forum 523/524. To do this, call up the **Configurator** and open the **Network: LAN** page. Click on the **Change** button.
3. Enter the current IP address configuration in **IP address** and **Network mask**. Confirm with **Apply**.
4. Configure the Forum 523/524's DHCP server to assign IP addresses. To do this, call up the **Configurator** and open the **Network: DHCP** page. Click on the **Change** button.
5. From **Status**, select the **address assignment** option. Confirm with **Apply**. The **DHCP** page is re-displayed.
6. Add the configured VoIP system telephones to the list of IP addresses. Click on the **New** button.
7. Enter the VoIP system telephone's **IP address** and **MAC address**. Enter the IP address reserved by the DHCP service programme. Confirm with **Apply**.

Restart the Forum 523/524 and all connected VoIP system telephones.

Tip: Configuring multiple VoIP terminals can be facilitated by using dynamic address assignment (please refer to *Dynamic Address Assignment for Specific Devices* starting on page 93).

Start Procedure

It may sometimes be useful to understand a VoIP system telephone's start procedure. Examples:

- A complex DHCP address assignment prevents the operation of the Forum 523/524's DHCP server within the LAN.
- A VoIP system should be operated with a non-broadcast-capable IP connection. This may be an RAS connection, a VPN connection or another type of routed connection.

An external DHCP server can also control a VoIP system telephone's start procedure. In this case, system software matching the type of VoIP system telephone must be transferred via TFTP.

The file name is determined by the telephone type.

Telephone type	File name
Forum IPhone 520	/ram/ip_tel/opi63.cnt
Forum IPhone 530	/ram/ip_tel/opi65.cnt
Forum IPhone 525 Forum IPhone 535	/ram/ip_tel/opi7x.cnt
Forum Base DECT IP	/ram/ip_tel/ip_rfp.cnt or /ram/ip_tel/ng_ip_rfp32.cnt
Forum Base DECT IP v2	/ram/ip_tel/ng_ip_rfp.cnt

After the VoIP system telephone has been connected to the mains power supply, the start procedure is as follows:

1. The boot loader starts and shows the VoIP system telephone's MAC address in the display. A DHCP request is sent simultaneously via broadcast on the 255.255.255.255 broadcast address.
2. An IP address, network mask and the default gateway for the start procedure are sent from the DHCP server. Via the "Next server" option, the DHCP server also provides the TFTP server's IP address and the operations software's file name. The DHCP server uses the MAC address to select the operations software file which matches the type of device.
3. The boot loader loads the approximately 2 MB operations software file from the specified TFTP server. The TFTP server's IP address and the file's name are shown in the display. The loaded operations software is started.
4. The operations software sends a DHCP request on the broadcast address 255.255.255.255. The VoIP system telephone now receives an IP address, network mask and default gateway for operations from the DHCP server. Using "Option 43", which is reserved for this purpose, the DHCP server also provides the IP address of the communications system and port number 8100 for registration.
5. The VoIP system telephone creates a TCP connection to the supplied IP-address/port-number combination and sends a registration query. The Forum 523/524 checks the MAC

address sent with the registration and confirms the registration request if the VoIP system telephone is listed in the menu **Telephony: Devices: VoIP Phones**. The keep-alive time, port number (8101) for telephony signalling and the value to use for the TOS byte are also communicated in the registration answer.

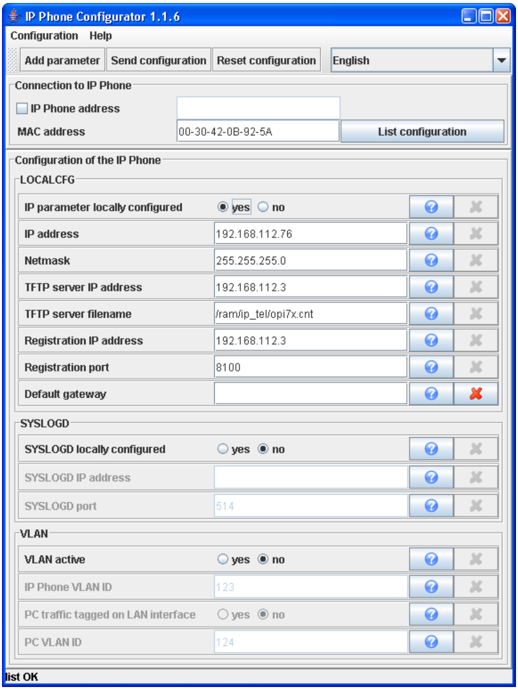
6. The VoIP system telephone creates a second TCP connection using the signalling port number 8101 and sends a registration.
7. Extra connections are created using the IP protocol RTP ("Realtime Transport Protocol") for call data when a call is created. For calls between two VoIP system telephones, port numbers above 8200 are used. For transmission to a Media Gateway card, a port in the range 1024 – 1087 is used.

If you wish to operate a VoIP system telephone via a routed IP connection (for example VPN or RAS) it may be necessary to configure an external DHCP server accordingly. Please note the selection of the codec and keep-alive time for RAS connections. This can be done by selecting the default profile **RAS** in the **Telephony: Devices: VoIP Phones** for the VoIP system telephone. The operations software provided via TFTP must match the type of device and communications system. You may also need to configure BOOTP, DHCP and TFTP servers for the VoIP system telephone.

Local Configuration

In addition to automatic configuration via BOOTP/DHCP, it is possible to manually configure a Forum IPhone 525 or a Forum IPhone 535. This can make sense, for example, when you wish to connect a VoIP system telephone at a distant location via router. This local configuration is saved permanently in the non-volatile memory of the VoIP system telephone. To change the local configuration, use an additional programme, the Java-based "IP Phone Configurator".

Note: *Java programmes can be run on all common operating systems. To execute Java-based programmes, you must install a suitable Java runtime environment on your operating system (JRE). This can be downloaded under the following web address: http://www.java.com/.*



1. The “IP Phone Configurator” can be started directly from the Product CD. Start Windows Explorer. Navigate to the Product CD. Double-click the “Forum\IpPhoneConfigurator.jar” file.

   The “IP Phone Configurator” dialogue opens. Select the desired language setting (“English” or “German”)from the drop-down menu.

2. Enter the network address of the VoIP system telephone. You have two connection types to choose from under **Connection to IP Phone**:

   – Deactivate the **IP Phone address** option to establish a broadcast connection via “UDP-Broadcast”. You have to select this type of connection if the VoIP system telephone has not

yet been assigned an IP address. IP broadcasts cannot be transmitted via router. The VoIP system telephone thus has to be directly connected to your PC via a hub or via a switch.

– Activate the **IP Phone address** option to establish a point-to-point connection via “UDP-Unicast”. Enter the IP address of the VoIP system telephone into the entry field. You can select this type of connection if the VoIP system telephone has already been assigned an IP address.

3. Enter the **MAC address** of the VoIP system telephone. You will find the MAC address on the underside of the device. Click on **List configuration**. The status bar at the bottom edge of the programme window displays “list OK”.
4. Change the desired settings under **Configuration of the IP Phone**. Click on **Reset configuration** to activate the standard settings for all entry fields.
5. Click on the **Send configuration** command to transfer the currently shown configuration to the VoIP system telephone. The status bar at the bottom edge of the programme window displays “send OK”.

Note: *The VoIP system telephone receives the configuration and sends a response. The new configuration is only saved and activated once this has happened. This can result in the “IP Phone Configurator” not receiving the response of the VoIP system telephone.*

Please note: *If you are operating multiple network cards with active IP configuration in your PC, this may mean that the loading of configuration data fails. First you deactivate additional network cards or use a point-to-point connection. Sending configuration data with a broadcast connection functions even without a response from the VoIP system telephone.*

You can implement the following settings:

IP parameter locally configured: Select the **yes** option to activate manual IP address configuration. Select the **no** option to activate automatic IP address configuration via BOOTP/DHCP.

IP address and **Netmask**: Enter an available IP address and the network mask to be used by the VoIP system telephone.

TFTP server IP address and **TFTP server filename**: Enter the IP address and the complete file name for the operating software of the VoIP system telephone (see table on page 132). For the Forum 523/524, use always the IP address of the communications system.

Registration IP address and **Registration port**: This is where you usually enter the IP address of the Forum 523/524 and the port number 8100.

Default gateway: Click on the **Add parameter** command to have this optional entry field displayed. Then enter the IP address of the router ("Default Gateway"). Click on the Delete button to remove the optional parameter.

SYSLOGD: For monitoring purposes, VoIP system telephone messages can be sent to a Syslog server. Activate the **yes** option and configure the **SYSLOGD IP address** and **SYSLOGD port** settings to activate this function.

VLAN (Expert option - do not change the "no" setting in standard cases): To improve transmission security or to enforce security guidelines, PC data transmission and VoIP data transmission can be separated using this method. Activate the **yes** option and enter the desired **IP Phone VLAN ID** for the VoIP data transmission. Enter a value ranging from 1-4094. Data will always be transmitted without a VLAN tag at the VoIP system telephone's PC access. If you activate the **yes** option for **PC traffic tagged on LAN interface**, PC data will be labelled with the **PC VLAN ID** at the LAN access. Please note that to change the VLAN settings, the VoIP system telephone has to be restarted.

Forum IPhone PC

Besides the hardware VoIP system telephones, PC software for VoIP telephony can also be deployed. This software can be used with the operating system Windows.

Software VoIP System telephone Forum IPhone 535 PC with one key extension

As well as VoIP system telephony from workstation computers, the Forum IPhone PC includes the following features:

- Usage via Mouse/PC keyboard
- “Drag & Drop” call number selection
- Integrated answering machine / recording function
- Terminal control for the sight-impaired
- Selectable user interface (“Skins”)
- Display language modification

The workstation computer requires a full-duplex-enabled sound card as well as a suitable headset for audio recording and play-back.

You will not need a licence to install Forum IPhone PC but you will require a licence to operate it with the Forum 523/524. Unit licences, enabling the simultaneous operation of a certain number of Forum IPhone PCs, are available.

The licences are activated in the Forum 523/524’s **Configurator** in the Menu **System: Licences**. The system software includes a licence for a demo version for temporary use (60 days). Please contact your Belgacom agent if you wish to purchase a permanent licence. Licences can be combined. Each licence can be activated only once.

Installation

Installation is done using a setup programme (“IPCsetup.exe”). The Forum IPhone PC can also be installed without a user interface. The programme can then be used via a CTI application (Net-TAPI or Forum CTI).

Start the installation programme Forum IPhone PC from the product CD and follow the installation assistant’s instructions.

Configuration

Analogue to the VoIP system telephones, the Forum IPhone PC creates multiple IP connections to the Forum 523/524. When you start the programme for the first time, the **Options** dialogue is automatically opened. Here you must configure the following values:

1. Enter in the **VoIP IP Address** field the Forum 523/524’s IP address.
2. Enter six hexadecimal-digits into the **Device ID** field. This device ID is not a MAC address, so overlapping with existent MAC addresses is possible. The device ID is configured in the **Configurator**, on the **Telephony: Devices: VoIP Phones** page.
3. Confirm with **OK**.

Notes

VoIP system telephony requires an active IP connection to a workstation computer. If a Firewall is installed for your workstation computer, you may need to explicitly allow this connection.

If you log on to the workstation computer using a different user name, you must reconfigure these values.

You can use any arbitrary sequence of digits not already in use in the LAN for the device ID. Select a random device ID to secure telephone usage. The device ID can only be read on the Web console.

The displayed menu texts and parts of the operations software are elements of the Forum IPhone PC installation, but they can be loaded from the Forum 523/524 via TFTP where necessary.

DECT over IP®

In order to achieve optimal network coverage, a DECT network with several DECT base stations can be operated. A DECT network is comprised of DECT terminals connected with the next respective base station (network cell). For users of a DECT terminal, the handover from DECT base station to base station is completely transparent. Even during a conversation, users are switched from one network cell to the next without any interruption. Administration of DECT terminals is done centrally via the Forum 523/524 Configurator in the **Telephony: Devices: DECT Phones** menu.

Note: *DECT over IP® is a registered trademark of Aastra Telecom Schweiz AG.*

Properties

DECT Base Stations

DECT base stations can be connected to the Forum 523/524 via U_{pn} accesses or via network (TCP/IP). These DECT base stations are available for the type of access selected:

U_{pn} DECT

- Forum Base DECT indoor v2: Access via U_{pn} with lines up to 1000 metres in length; integrated antennas; 4 voice channels (8 when using 2 U_{pn} accesses)
- Forum Base DECT outdoor v2: like the Forum Base DECT indoor v2; mounted outside enclosed areas (IP55); external antennas

Note: *The newer DECT base stations Forum Base DECT indoor v2 and Forum Base DECT outdoor v2 can be simultaneously operated with the older DECT base stations, Forum Base 500. Fax transmissions (group 3 with ECM) and SARI (roaming with Secondary Access Rights Identification Broadcasts) can be done using the newer DECT base stations. Data transmission via DECT is not available with the newer DECT base stations.*

IP-DECT

- Forum Base DECT IP indoor: Access via shielded CAT5 Ethernet cable (STP cable, Shielded Twisted Pair cable) with up to 100 metres of cable from the last Ethernet switch, integrated antennas; 8 voice channels
- Forum Base DECT IP outdoor: like the Forum Base DECT IP indoor; mounted outside enclosed areas (IP55); external antennas
- Forum Base DECT IP-WLAN indoor: Access via shielded CAT5 Ethernet cable (STP cable, Shielded Twisted Pair cable); offers simultaneous function of a WLAN Access Point conforming with the IEEE 802.11b/g protocol; external antennas; 8 voice channels

When started, the operating software for the DECT over IP base stations is transmitted via TFTP protocol from the Forum 523/524. The configuration for the start sequence is transmitted by the DHCP server of the Forum 523/524 to a DECT over IP base station for the start sequence.

IP-DECT (NG / New)

- Forum Base DECT IP indoor v2: Access via shielded CAT5 Ethernet cable (STP cable, Shielded Twisted Pair cable) with up to 100 metres of cable from the last Ethernet switch, integrated antennas; 8 voice channels
- Forum Base DECT IP outdoor v2: like the Forum Base DECT IP indoor v2; mounted outside enclosed areas (IP55); external antennas
- Forum Base DECT IP-WLAN indoor v2: Access via shielded CAT5 Ethernet cable (STP cable, Shielded Twisted Pair cable); offers simultaneous function of a WLAN Access Point conforming with the IEEE 802.11a/b/g/n protocol; external antennas; 8 voice channels

The operating software for these DECT over IP base stations is transferred from the Forum 523/524 at the first-time start with the TFTP protocol. For later starts, the operating software in the memory of the device is used. The configuration for the start sequence is transmitted by the DHCP server of the Forum 523/524 to a DECT over IP base station for the start sequence.

Note: *You can simultaneously use older and newer DECT over IP base stations. In this case you have to select a newer DECT over IP base station Forum Base DECT IP v2 as DECT over IP Manager.*

If VoIP telephony is already being used, Ethernet cable access makes good sense. Transmission of telephony signalling and voice data via TCP/IP also offers usage of existing network infrastructure and an increase in range using suitable methods. VPN connections, for example, can be used for data links to provide service to remote or hard-to-reach locations.

Note: *The DECT base stations Forum Base DECT IP indoor (v1 & v2), Forum Base DECT IP outdoor (v1 & v2) and Forum Base DECT IP-WLAN indoor (v1 & v2) support the DECT encryption function. This feature is however, only available if all the DECT base stations support it.*

Features

All DECT over IP base stations can be connected to a CAT5 Ethernet cable with a 10/100 Base T. Power is supplied either via Power-over-LAN (IEEE 802.3af) or via an additional power supply unit. A base station of the type Forum Base DECT IP outdoor can only be supplied via PoE.

Please note: *The WLAN function of the Forum Base DECT IP-WLAN indoor is activated only when connected to the 100 Base T.*

DECT terminals offer all system telephony features. DECT telephones supporting the GAP standard can also be operated. Transparent GAP device handovers are supported. DECT encryption of calls can be deactivated for the Forum Base DECT IP (v1 & v2) if desired.

VoIP audio communication between the DECT over IP base station and the Forum 523/524 is made via the RTP/RTCP protocol. RTP

voice data are directly converted into DECT voice data by the base station. The base stations support the following VoIP codecs:

G.711	uncompressed
G.723	compressed
G.729	compressed

Configuration

One of the DECT over IP base stations that is installed assumes coordination and configuration of the DECT over IP functions ("DECT over IP Manager", OMM). Select a base station that has a dependable data link to the Forum 523/524.

Note: *You can determine a second base station as additional DECT over IP Manager ("Standby Device"). If the first DECT over IP Manager fails, the second base station takes over this critical function after some minutes and a reset of the DECT network.*

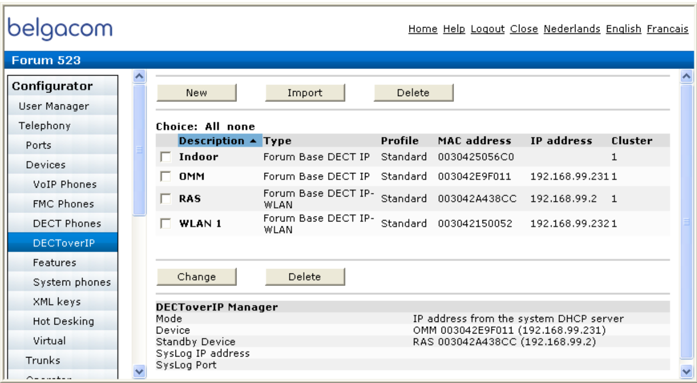
Go to the **Telephony: Devices: DECT over IP** page in the **Configurator**. Click on **New**, to add a DECT over IP base station. Enter the **MAC address** of the base station that you have selected to be the DECT over IP Manager. The MAC address of the base station is located on its type label. Enter an **IP address** for this base station. Confirm with **Apply**. Now click **Change**, to determine the DECT over IP Manager. In standard cases, leave the **Mode** in the "IP address from the system DHCP server" setting. Select the desired DECT over IP base station with the **MAC address (IP address)** setting. Confirm with **Apply**.

If the Forum 523/524's DHCP server is not configured for static address assignment, you first need to configure the IP address of the DECT over IP Manager with the help of an additional programme (see *Local IP Address Configuration* starting on page 149). Also change the **Mode** setting on the **Telephony: Devices: DECT over IP** page to "IP address configured local" and enter there the configured IP address for the DECT over IP Manager. All other base stations can be operated using either a fixed IP address or an IP address assigned dynamically via DHCP.

Please refer to the information given in the chapter entitled *LAN DHCP Server* starting on page 130.

Note: *A base station cannot be operated as a DECT over IP Manager and a WLAN-Access Point simultaneously.You should therefore use a DECT over IP base station which does not have WLAN function as your DECT over IP Manager.*

Create a separate entry for each DECT over IP base station and for the DECT over IP Manager on the **Telephony: Devices: DECT over IP** page. You use these entries to determine the VoIP data compression (“Profile”).



Configurator: Telephony: Devices: DECT over IP

User administration and set-up of DECT terminals is done in the Configurator of the Forum 523/524 as well.

The DECT over IP Manager offers a separate web user interface to manage the settings of devices with WLAN functions. Therefore at least one WLAN-RFP has to be configured. If everything is configured correctly, you will see the **WLAN Config** link. Login as the user “Administrator” with the currently set administrator’s password of the Forum 523/524.

Dual Operation

Simultaneous operation of base stations via U_{pn} access and base stations via Ethernet access is possible with the Forum 523/524. Transparent handovers, for example, are only possible when between DECT base stations using the same access technology.

When switching over to a DECT base station using different access technology, a DECT terminal automatically re-establishes a connection (roaming).

Synchronisation

Transmissions of all DECT base stations at a single location must be synchronised in order that DECT terminals are able to receive multiple DECT base stations simultaneously. Synchronisation can be conducted via U_{pn} access. It cannot be conducted via an Ethernet/IP connection. DECT over IP base stations are thus synchronised via wireless connection.

When planning a larger sized DECT network, it is advisable to take the following points into consideration:

- All DECT over IP base stations at a single location must be able to receive at least one, or even better, two neighbouring base stations. Synchronisation requires less signal strength than a voice connection does.
- Synchronisation range is increased using multiple base stations. To decrease the probability of a connection breakdown, base stations should not be arranged in chain formation. The signal should be distributed with a network that is as extensive as possible and where each base station is supported by multiple synchronisation partners.
- To re-synchronise, first wait for all current connections to be terminated.

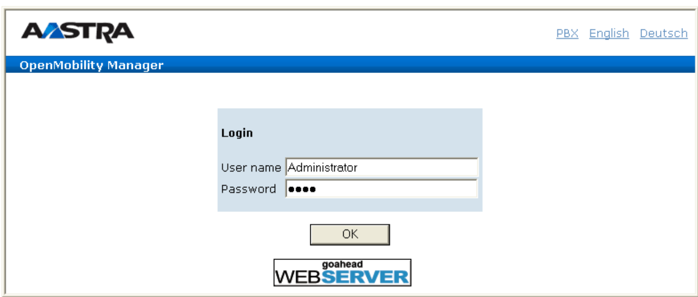
You can operate a DECT network consisting of several remote locations (“clusters”). A cluster is a number of DECT base stations that operate synchronously with each other. No handover is possible between DECT base stations from different clusters. You should configure a second cluster for DECT base stations of a second location.

Setting up the WLAN Function

The Forum Base DECT IP-WLAN indoor (v1 & v2) DECT over IP base station provides the additional function of a Wireless LAN Access Point (WLAN-AP). WLAN refers to data transfer by means

of radio waves in accordance with the IEEE 802.11b/g standard. This standard enables a wireless connection to be made to an Ethernet network (LAN) using suitably equipped user terminals. Data transfer via radio waves is very fast. Depending on the conditions of the operating environment, it can reach speeds of up to 54Mbit/s (gross).

WLAN settings are configured centrally for all Access Points using a separate Web configurator, which can be found at the IP address of the DECT over IP Manager. You can reach this address by entering the IP address of the DECT over IP Manager directly into the address bar of your Web Browser. Alternatively, you can also go to the **Configurator**, to the page **Telephony: Devices: DECT over IP** and click on the **WLAN Config.** button. Log in under the **User Name** “Administrator” and enter the same password as for the Forum 523/524.



DECT over IP/OpenMobility Managers Login Page

The WLAN function and the function of the DECT over IP Manager cannot be used simultaneously on the same DECT over IP base station, so you will always need at least two DECT over IP base stations. The WLAN settings are then made as follows:

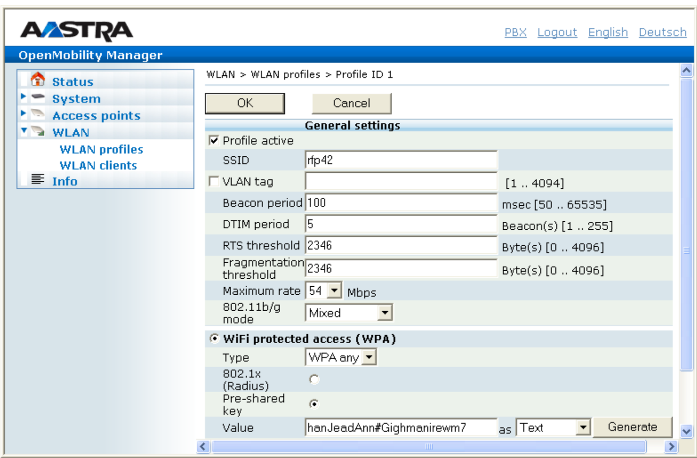
1. Set up the existing DECT over IP base stations in theForum 523/524’s **Configurator**. Go to the DECT over IP Manager’s Web Configurator.
2. On the **WLAN Profiles** page, configure at least one set of settings (see below under: *Setting up a WLAN Profile*). Note down the password you have used (“Pre-Shared Key”), so that you will be able to use it again later when setting up wireless user terminals or notebooks.

3. Assign the desired WLAN profile on the **Radio Fixed Parts** page. Click on spanner symbol () on the left next to the relevant DECT over IP base station. Under **WLAN Settings** select the number of the configured **WLAN Profile**. Confirm your settings with **OK**. You can use one profile for multiple DECT over IP base stations.

You can now use the WLAN function of your WLAN-enabled DECT over IP base stations and set up the user terminals as required.

Setting up a WLAN Profile

The WLAN function of the Forum Base DECT IP-WLAN indoor (v1 & v2) DECT over IP base station also includes such rarely-required features as networks for large company premises or airports. In this guide we will, for the sake of brevity and clarity, describe only those features required for secure standard operation.



DECT over IP/OpenMobility Manager: WLAN Profiles

Use the following settings for standard operations.

General Settings

- Select the desired **WLAN Profile** and activate the **Profile Active** option.
- Enter a **SSID** (Service Set Identifier, wireless network identification) to identify a network. This network identification

is transmitted at regular intervals, making it easier to find the networks you're looking for, using the "View available wireless networks" function in Windows XP, for example.

- For standard operation you should leave the following settings at their default values: **VLAN Tag** at 0 (Off), **Beacon Period** at 100 ms, **DTIM Period** at 5, **RTS Threshold** at 2347 (Off), **Fragmentation Threshold** at 2346 (Off), **Maximum Bitrate** at 54 Mbit/s, **802.11b/g Mode** at "Mixed" and **Interference Avoidance** on "Off".

Tip: If you are using only modern WLAN cards with 802.11g, you can further speed up data transfer by configuring the setting **802.11b/g Mode** to "802.11g only".

- You can prevent the transmission of wireless network identification (SSID) with the **Hidden SSID Mode** setting. This will however make network identification difficult and does not generally increase data security, so it is preferable to leave this on the default setting of "Off".

Security Settings

On no account should you use **Open System** or **Wired Equivalent Privacy (WEP)** settings**,** whether out of convenience or in order to avoid configuration problems, unless of course you want to start up an Internet Cafe!

- Activate the **Wifi Protected Access (WPA)** option.
- Under **Type** select the "WPA v.1"setting. If you are running the Microsoft Windows XP operating system from ServicePack2 or higher on your computer, you can use the "WPA v.2" setting.
- For standard operation select the **Pre-Shared Key** option. Enter a password in the **Value** input field and leave it set to **Text**. Use a password with the following characteristics:

  – No words or names that can be found in a dictionary

  – At least 8 characters long

  – It should also include numbers, a mixture of upper and lower case and special characters

You could also use the **Generate** button to generate a password. Some WLAN configuration software does not convert text into hexadecimal values as a standard procedure. If this is the case, go to the **Hex Value** setting and select the **Generate** button.

- Leave the **Cipher Length** setting at 256 Bit and the **Distribution Interval** setting at 120 seconds. You will not usually need the settings for **WME** or for configuring **Multiple SSIDs** for standard operation.

Tip: If you are running an Internet-Cafe without using powerful encryption, you should, for the sake of your customers' security, prevent them from being able to access each others' computers. Activate the **BSS Isolation** option. You can also stop unpleasant guests from using the system with a **MAC Address Filter** – but this will not hold up users who know about this function for long.

Configuring for a Remote Location

If you are using a DECT over IP base station in the same LAN as the Forum 523/524, the IP address configuration and software loading procedure which are run when a DECT over IP base station is started are handled by the Forum 523/524 using the DHCP and TFTP protocols.

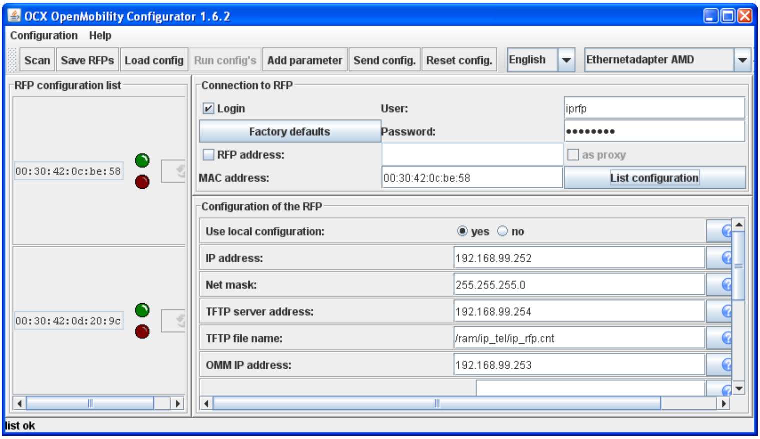
For the DHCP function to be available, the DECT over IP base station must be able to reach the Forum 523/524 via a "Broadcast". In the case of a remote location this kind of access - via a VPN connection for example - will not be possible. As in the case of an IP system telephone, you will have to acquire the necessary system software for the DECT over IP base station with the help of a TFTP server.

Local IP Address Configuration

The IP address configuration can be set up as a "local configuration" with the help of an additional programme.

1. Call up Windows explorer. Browse to the communications system's product CD. In the "Forum" directory, double-click the "OM_Configurator.jar" file.

Note: *You need a Java runtime library to be able to run Java programmes. If you don't have one installed, you can download the installation programme from the http://www.java.com/ web site.*



OpenMobility Configurator

2. To log in to the dialogue, you enter:

   User: "admin"

   Password: "Admin"

3. Enter the **MAC address** of the DECT over IP base station. The MAC address will be printed on the label on the DECT over IP base station's casing. Click on **List configuration**.

   The DECT over IP base station's current configuration will be displayed.

4. Change the DECT over IP base station's IP address configuration. Activate the **Use local configuration** option ("yes") and enter the required details:

   – **IP address**: Static IP address of the DECT over IP base station

   – **Net mask**: Subnet mask of the DECT over IP base station

   – **OMM IP address**: IP address of the DECT over IP Manager. For the actual DECT over IP Manager simply repeat the entry from the **IP address** entry field.

   – **OMM port number;** Leave the default setting on "16321".

– **PBX IP address**: IP address of the Forum 523/524

– **PBX port**: Leave the default setting on “8099”.

5. Under **TFTP server address** enter the IP address the operating software is to be downloaded from. This will usually be the communications system’s IP address. Leave the **TFTP file name** setting on the default setting (“/ram/ip_tel/ip_rfp.cnt”).
6. For a remote location, the Forum 523/524’s LAN will usually be accessed via a (VPN) router. Click on **Router addresses**: **[+]**. Enter the router’s IP address (“default gateway”). Confirm by clicking on **Add**.
7. Click on **Send config** to activate the desired IP address configuration.

PBX Networking

Forum 523/524 provides all the features necessary for PBX networking. You need PBX networking in the following cases:

- To operate the Forum 523/524 as a subsidiary system on another PBX. This will also allow you to use the Forum 523/524 as a DECT server, for example.
- To network several Forum 523/524s into a PBX system.
- To use flexible configuration possibilities of trunk lines for a Forum 523/524.

All settings that affect the configuration of PBX networking can be found in the Configurator menu **Telephony: Trunks** and in the **Telephony: Settings** dialogue under **QSIG linking**. Refer also refer to the corresponding help topics in the Forum 523/524 online help.

You can use ISDN point-to-point connections (Q.SIG or DSS1 protocol) or IP connections (Q.SIG-IP or SIP tie line protocol) for TK system networking.

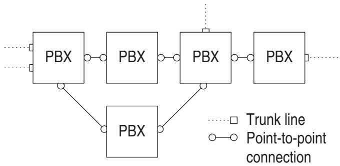
Note: *If you do not need the features of PBX networking, the simplified configuration is sufficient in most cases. For this purpose, assign the preconfigured bundles (bundles)* ***Multi-terminal access*** *or* ***System access*** *to the ports. The preconfigured route called* ***External trunk*** *now makes it possible to seize an external line immediately or by first dialling the prefix “0”. You can rename the preconfigured bundle and the preconfigured route if required, but you cannot delete them.*

Connections

Networking two or more TK systems means interconnecting them. The Forum 523/524 allows you to use the following connections:

- ISDN trunk lines
- ISDN point-to-point connections (Q.SIG) on external S_0 ports
- IP network connections (Q.SIG-IP or SIP tie line)

Note: *Networking via IP network connections requires a MGW interface card Forum 525 Media Gateway Card.*



Example of a PBX network

Various line types and transmission protocols can be used for point-to-point connections. The required network topology (distance, connection capacity) determines which type of point-to-point connection is most suitable.

An IP network connection with Q.SIG-IP or SIP tie line can be established either with a direct LAN connection or via a fast VPN connection (see also *Branch Link* starting on page 97). When the connection is via internet, encryption is advisable for security reasons.

Point-to-Point Connection Technology

Protocol: Q.SIG or DSS1

The Q.SIG protocol, designed for ISDN point-to-point connections, is the preferable choice as the transmission protocol; alternatively, the DSS1 protocol, designed for ISDN dial-up connections in the Euro-ISDN, can be used. Certain PBX networking features can only be used with the Q.SIG protocol, however. In particular, the connection designation as internal or external call and also the name of the caller cannot be transmitted via the DSS1 protocol.

Both protocols implement communication on several protocol layers:

- L1: Layer 1 defines the physical line properties and the electrical coding of signals.
- L2: Layer 2 enables communication via individual error-protected channels that are independent of each other.
- L3: Layer 3 defines the administration of the individual channels and implements the features designed for ISDN.

Master/Slave

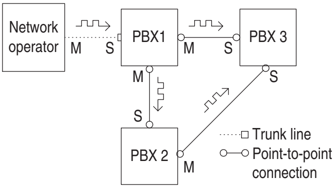
For an ISDN connection, it is possible to determine which PBX is the protocol master and which the protocol slave. This relationship can be determined for all three protocol layers independently of one another.

For each protocol layer, the PBX at the other end always has to be suitably configured. If one PBX is the protocol master for a layer, the other PBX must be the protocol slave for this same layer. Normally all three protocol layers are configured identically. In the case of a trunk line, the network operator is the protocol master for all three layers.

L1 Clock

To enable PBXs in the ISDN network to communicate with each other, they must be “clock-aligned”. The L1 protocol master sets the clock for layer 1, and the L1 protocol slave adopts (synchronises to) this clock.

When planning a PBX networking scheme, you must make sure that the L1 clock propagates from a master via a number of PBXs.



Example: propagation of the L1 clock

If more than one port with the setting **L1 Type** = “Slave” is configured on an Forum 523/524 and the setting **L1 sync possible** has been activated, then one of the ports is automatically defined as the L1 clock source. The Forum 523/524 will automatically switch the clock source to another port configured as an L1 clock source (if a line fails, for example).

Please note: *Reciprocal or circular application of the L1 clock is not allowed.*

Example: In the above case you could reverse the L1 slave/master setting for the connection between PBX 1 and PBX 3. However, if you then activate the setting **L1 sync possible** for the port of PBX 1, this may cause parts of the PBX network to stop functioning temporarily.

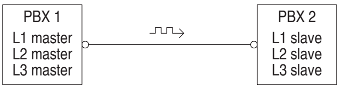
When applying the L1 clock of trunk lines, you can assume that the public network is “clock-aligned”. So, in the above example, you can connect additional trunk lines to one of the PBXs.

Point-to-Point Connection Lines

There are different types of lines available for an point-to-point connection between two PBXs, depending on the distance between them.

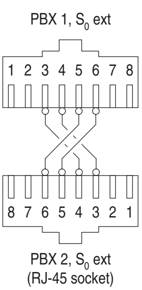
Direct Connection

This type of ISDN point-to-point connection joins the two systems directly to each other using a crossover twisted-pair cable. An S_0 connection can be used for distances up to 1,000 metres. Normally one PBX is the protocol master for all three layers, and the other PBX is the protocol slave for all three layers.



Direct connection

Use the RJ45 jacks on one of the external S_0 ports for an S_0 connection between two Forum 523/524s. You can use the corresponding pressure terminals for S_0 ports on interface cards.



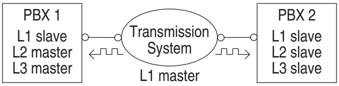
Wiring of a direct connection

Note: *If you use an S_0 port on an interface card (pressure terminal) and an S_0 port with an RJ45 jack for the direct connection, make sure you make the necessary changes to the port assignment (see S_0 Ports starting on page 48).*

The external S_0-ports of a gateway can also be used for a direct connection.

Connection via an Active Transmission System

For distances exceeding the range of a direct connection, an active transmission system can increase the range to up to 50 km. Normally the L1 master is the transmission system for the two connected PBXs. For the protocol layers L2 and L3, one PBX is normally the protocol master and the other PBX is the protocol slave.

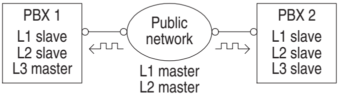


Connection by an active transmission system

Note: *The active transmission system itself gets its L1 clock either from the network operator or from a clock generator connected by wire.*

Connection via the Public Network

Point-to-point connections via the public network of a network operator can be used for bridging distances beyond 50 km. Due to the long distance involved, for technical reasons it is not possible to synchronise the L2 protocol. Consequently, the public network is normally the protocol master for protocol layers L1 and L2. One PBX is therefore the L3 master and the other PBX the L3 slave.



Point-to-point connection via a public network

IP Network Connections

If you are operating a fast and continuous network or internet connection at two or more sites, you can also network the PBX systems via a network connection.

There are two different protocols available for networking TC systems via network connections: Q.SIG-IP and SIP tie line. The vari-

ous features of these protocols are summarised in the following table:

TC system networking via network connections

Feature	**Q.SIG-IP**	**SIP tie line**
Establishing a call (“basic call”) with call-number transmission	yes	yes
name information transmission	yes	yes
single number dial possible	yes	no
transparent codec interconnection	no	yes
automatic feature detection	no	yes

Connection via Q.SIG.IP

The Forum 523/524 communications system supports transmission of the Q.SIG protocol – intended for ISDN point-to-point connections – via network connection (Q.SIG-IP). Protocol and call data are exchanged using IP connections with Q.SIG-IP.

The number of simultaneous conversations possible will depend on the capacity of the network or internet connection and the compression method used. A multiple S_{2M} point-to-point connection is simulated for each Q.SIG-IP bundle. This means that 5 virtual D channels and up to 120 voice channels are available. The Media Gateway Card channels are used for Q.SIG-IP (see *MGW Interface Card* starting on page 116). Q.SIG-IP connection data are subject to codec compression (please refer to *Voice over IP (VoIP)* chapter regarding *Fundamentals* starting on page 107). Q.SIG-IP also transfers the voice data directly from terminal to terminal via the RTP protocol. In certain cases, for example, when an incoming external call is placed via multiple TK systems, one or more RTP proxies may be used to forward the connection.

Currently, there are no standards for the necessary extensions to the Q.SIG protocol. This means that you can only use Q.SIG-IP between Forum systems.

Networking two Forum 523/524 systems using Q.SIG-IP requires 2 licences – one licence per system. The number of possible voice connections is not restricted by the licence.

Go to the **Telephony**: **Trunks**: **Trunk group** page in the **Configurator** to set up a Q.SIG-IP connection. Create a new bundle and select the **Access type** “System Access”. Select “Q.SIG-IP” under **Protocol**. Configure the IP address of the other system, the port numbers to be used, the number of possible voice connections. Select a VoIP profile for the codec selection. Please refer to the relevant help topics in the Online Help for the Forum 523/524 as well.

Note: *Q.SIG-IP cannot be operated using a connection with NAT. For a Q.SIG-IP connection, a branch connection or another VPN connection is required.*

Connection via SIP tie line

The Forum 523/524 communications system supports connections using a SIP tie line for TC system networking. A SIP tie line is a SIP line which requires no login which can establish multiple call connections simultaneously. No SIP provider is required for establishing the connection via SIP tie line.

The number of simultaneous calls possible depends on the network or internet connection capacity and the compression procedure being used. The channels of a media gateway card are used for a SIP tie line (see *MGW Interface Card* starting on page 116). The data of a SIP tie line connection is subjected to codec compression (see under *Fundamentals* starting on page 107 in the chapter *Voice over IP (VoIP)*). Call data is transmitted directly from terminal to terminal via the RTP protocol with SIP tie line as well. In certain cases, for example when an incoming external call is switched via multiple TC systems, there may be one or multiple RTP proxies involved.

One of the special features of a SIP tie line connection is using transparent codec interconnecting, for example to make use of HQ audio or video telephony with appropriate terminals (see *Transparent codec interconnection* starting on page 113). In addition, you can find out what features are supported by the other station

via the SIP tie line protocol. This makes it possible to automatically adapt to the respective other station.

Currently, there is no generally accepted standard for the protocol used with a SIP tie line. This means via SIP tie line you can only use connections between Forum systems.

Two licences are required when networking two Forum 523/524 with a SIP tie line – a licence for each end point. The number of possible call connections is not restricted by the licence.

Open the **Telephony**: **Trunks**: **Trunk group** page in the **Configurator** to configure a connection via SIP tie line. Create a new bundle and select **Access Type** “System access”. Select “SIP Tie-Line” under **Protocol**. Configure the IP address of the other system, the port number to be used (the same port number at both end points), the number of possible call connections. Select a VoIP profile for codec selection. Please note the corresponding help topics in the online help Forum 523/524.

En-bloc dialling only is supported with a SIP tie line as with other SIP connections. To establish a call connection you have to first end call number entry with the hash key or wait a certain length of time. This length of time can be defined in the **Time to ready dial out** input field when configuring the SIP tie line (3-5 seconds are usual). In addition, you can activate a cache for accelerated dial out conclusion for the call numbers most recently dialled with the **Dial out cache** option.

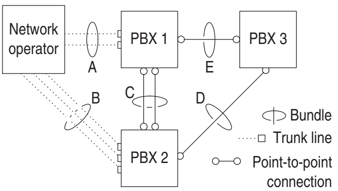
Note: *A SIP tie line cannot be conducted via a NAT connection. A branch connection or another VPN connection is necessary for a connection via SIP tie line.*

Configuration

The possible configurations described below can be set up in the Web console using the **Telephony: Trunks** menu.

Bundles

A **bundle** is a group of lines of the same type and direction. A line can only be assigned to one bundle.



Example of a PBX network with bundles

In the above example, the following bundles are configured for PBX 1:

- Two S_0 lines in a multi-terminal configuration to the network operator which are assigned to the “A” bundle.
- Two S_0 point-to-point connections to PBX 2 which are assigned to the “C” bundle.
- One S_0 point-to-point connection to PBX 3 which is assigned to the “E” bundle.

Note: *A line or a bundle cannot be seized directly. It is always performed indirectly via a route.*

Routes

A **route** is a group of bundles enabling a connection in one direction. If the first bundle of a route is fully utilized, the next bundle is seized (“bundle overflow”). One bundle can also be used for different routes.

In the above example, a route set up for PBX 1 allows a connection to PBX 2. Bundle “C,” “E” and “A” are assigned to this route. If a user connected to PBX 1 wants to reach a party in PBX 2, lines will be seized in the following order:

- PBX 1 first searches for a free channel in the “C” bundle.
- If all the lines in bundle “C” are busy, the system tries to set up a connection via bundle “E”. PBX 3 switches the connection through, provided it is appropriately configured (refer to *Numbering* starting on page 162).
- If it was not possible to set up an indirect connection via PBX 3, the system tries again via bundle “A”. The “prefix” necessary for this can be configured with the route.
- The user does not get a busy signal until the attempt to set up an indirect connection via the network operator has also failed.

Note: *If an internal connection is switched via a network operator, the call is signalled using the external number of the calling PBX.*

For each route you can define a randomly selectable code digit for seizing the route. You can also configure whether a user is authorised to seize a particular route, whether LCR is to be used for one of the bundle and the criteria (business or private call, booking numbers) for evaluating call data.

Numbering

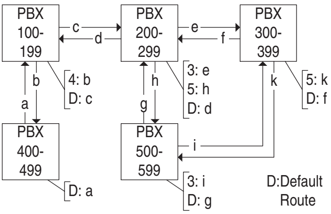
A user can seize a particular route by pre-dialling a specific code digit. With this “open numbering”, a user must always dial this code digit and then the telephone number in order to reach a party in another PBX.

If none of the telephone numbers in your PBX network occur twice, you can also configure “closed numbering”, allowing the same telephone number to be used for reaching each user within the PBX network.

With closed numbering, the Forum 523/524 determines which route to seize from the telephone number dialled. The information needed for routing a call can be configured in a numbering table containing up to 100 entries. You use this table to assign telephone numbers and/or ranges of telephone numbers to a particular route.

A **default** entry in the numbering table makes it possible to seize a “default route” for all remaining unassigned numbers. In partic-

ular, this simplifies configuration of the Forum 523/524 as a subsidiary system: the only entry you assign to the **default** entry is the route to the host system



Example of closed numbering tables

The automatic switching of call requests (i.e. routing) by means of bundle overflow or default numbering can lead to “circular switching”.

To avoid this, a “transit” counter is incremented whenever a connection is switched through on Q.SIG lines. When the configured maximum value is reached, further switching stops.

Technical Details

A different PBX number must be set for each Forum 523/524 in a PBX network. This setting can be found in the Web console, in the menu **Telephony: Settings** under the heading **QSIG linking**. You can also set the maximum value for the transit counter there. This value depends on the topology of the PBX network and should allow the system to have the maximum number of further connections possible.

You can display the connection status of the lines at any time in the Configurator menu **System Info: Telephony: Trunks**. You should check this in particular after making changes to a configuration to see whether all the lines used for system networking are operable.

Some of the features possible in Q.SIG are not supported by Forum 523/524 with all their options, for example callback on busy within the Q.SIG network. The call categories defined in Q.SIG (e.g. Emergency Call, Operator, Normal) and the Q.SIG name transmission feature (“user names”) are fully supported.

The code digits to be used for seizing a route with open numbering are not transmitted to the destination PBX and thus cannot be evaluated by it. To reseize a route (for example for a callback), you must set the appropriate digit prefixes in the bundle configuration for the routes to be reseized.

Tip: If, for example, you are configuring a route which can be seized using routing code "5" and have selected one or more bundles for this route, change the **Prefix for dest. call number at incoming internal** setting to "5" for this bundle in order to enable the route to be reseized.

Due to their hardware properties, not all S_0 ports of the Forum 523/524 can be fully used for TK system networking. Depending on the type of system, some ports can only be operated in the L1-Master mode or the L1-Slave mode. The external S_0 ports can be set according to the following table.

	S_0 1	**S_0 2**
Forum 523/524	S	M/S

Legend

S = Slave
M/S = Master/Slave

Note: *The S_0 ports on extension cards can be operated in both the L1-Master mode as well as the L1-Slave mode.*

Telephony

E.164 conversion

The Forum 523/524 communications system supports two different types of call numbers when dialling external call numbers. Usually you enter the code for a route, e. g. a “0” for the “external line” route. Then you enter an external call number. The external call number can be either a local area code or a country area code.

The additionally configurable “E.164 conversion” feature enables you to enter the entire international call number. Using the international call number makes sense in the following applications:

- When using the “Fixed Mobile Conversion” (FMC) feature as the call number is usually dialled from the local directory of the mobile telephone being used.
- When dialling via a computer programme connected with TAPI where call numbers are often already in the international format due to synchronisation with a mobile telephone.
- When importing or comparing directory data when directory entries are in the international format.
- When networking communications systems with locations in different local areas or in different countries.

The “E.164 conversion” analyses an international call number. The analysis divides the call number into multiple parts: the international area code, the local area code, the access call number and if necessary the extension. The call number is respectively abbreviated, eliminating any unnecessary area codes. The abbreviated call number is then used, e. g. to execute a call.

Configuration

You can configure the “E.164 conversion” feature for each bundle separately. This is possible for point-to-point configured bundles (“system access”), for point-to-multi-point configuration (“multi-terminal access”) and for SIP trunks:

1. Call the Forum 523/524 communications system **Configurator**. On the introductory page, change the **Level** option to **Expert**.
2. This step is optional as the country code is preallocated due to the **Country** setting under **System**: **Common**.

   Call the **Telephony**: **Settings** page. Click on the **Change** button. Enter the country code without a zero in front into the **International area code** field, e. g. “49” for Germany. The setting in the **Own area code** field is not relevant for the “E.164 conversion” feature. Confirm with the **Apply** button.
3. Call the **Telephony**: **Trunks**: **Trunk group** page or the **Telephony**: **Trunks**: **SIP trunks** page. Click on the desired bundle or desired SIP trunk. Activate the **E.164 conversion** option.

   The following setting is only relevant if the local area code is not part of the ISDN-MSN (Germany and Austria) in your national ISDN. If you have selected “Germany” or “Austria” under **System**: **Common** as the **Country** setting, you also have to enter the local area code into the **Area code** field for the bundle. The prefixed zero is not necessary.

   Confirm with the **Apply** button.
4. Call up the **Telephony**: **Trunks**: **Route** page. Check whether the **Type** setting for the used routes is set to “Private” or “Business”. For routes of the type “Internal” the “E.164 conversion” is not active.

The Q.SIG bundle used for networking communications systems cannot be used with the “E.164 conversion” feature. Please keep in mind that the differentiation between access call number and extension only takes place with system access or DDI-capable SIP trunks and also only when using direct extensions. With a call number allocated via call distribution there is no automatic differentiation between external and internal call numbers.

Note: *For incoming calls via a bundle with the "E.164 conversion" feature each external call number ("CLIP") appears converted into the international format and is also saved in this format e. g. in the caller list. Keep this in mind when entering call numbers used for authenticating (CLIP-Auth).*

Example

The following example explains the "E.164 conversion" function on a terminal which is operated on a system access with the following configuration:

Attribute	Number
International area code (Country Code, CC)	32 (Belgium)
Local area code (National Destination Code, NDC)	2 (Brussels)
Access call number (Subscriber Number, SN)	4444 (company in Brussels)
Extension or internal call number (Direct Dialling, DDI)	999 (company extension)
Code for the "External trunk" route	0

Various call numbers are now dialled from this terminal:

Number dialled	Number actually used
0003311234567 (foreign country, Paris)	003311234567 The code for the route is not transmitted via ISDN.
00032501234567 (domestic, Bruges)	0501234567 The international area code is replaced with a "0".
0003221234567 (domestic, Brussels)	021234567 The international area code is replaced with a "0". One's own local area code is not removed.

Number dialled	Number actually used
0003224444888 (Internal)	888 The international area code, the local area code and the access call number are deleted. The destination is called internally without using the ISDN connector.
+3224444888	888 You can only enter the plus sign (after E.123) with a SIP telephone (see *Internal SIP Subscribers* starting on page 119). There is a “E.164 conversion” in this case also.
0024444888	024444888 There is no “E.164 conversion” without an international area code.

Further Information

Please note the following information when using the “E.164 conversion” feature:

- When all external lines are occupied, the “Congested” state is only indicated later on during the dialling process.
- Emergency calls are always executed without “E.164 conversion”.
- Call number assignments in the call distribution are not evaluated for the “E.164 conversion” feature. For an assigned MSN, e. g. no automatic internal dial is executed even when the destination could be reached this way.
- Depending on the telephony provider, you can possibly also use the area code of your own country without “E.164 conversion”. Differentiating between external and internal call numbers can only be done using the “E.164 conversion”.

Number Handling with E.164 and Lists – Black List, White List, etc.

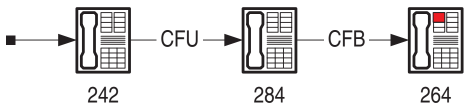
The combination of E.164 numbering scheme and black-, white, special lists requires explicit duplex/triplex handling of numbers in all eventually possible formats.

> Example: 0-6104-1234, 0-030-6104-1234, 0-0049 30-6104-1234.

All these possible formats can be dialled and so work around the e.g. black list. That has to be considered in programming these features.

Call Forwarding

The current version of the Forum 523/524 communications system also offers configuration of multi-level call forwarding. When you forward a call number that has already been forwarded, this results in a call forwarding chain.



Call Forwarding Chain

Multiple call forwarding is executed independent of the call forwarding type. An overview of the various call forwarding possibilities is nonetheless helpful for the following explanations.

Call Forwarding

Name	Description
Call forwarding immediately (CFU)	Immediate and unconditional call forwarding
Call forwarding on busy (CFB)	Call is forwarded only if user is busy
Call forwarding after time (CFNR)	Call forwarding is only executed after a definable time interval
Call Diversion (CD)	Is manually executed upon an incoming call from the user

Call Forwarding

Name	Description
Virtual call number	A virtual call number is always diverted to a destination call number
Call forwarding of a hunt group	Users of a hunt group can also configure respective call forwarding
Call forwarding to external	Call forwarding to an external call number or via remote-controlled dialling (Call Through)
Call forwarding by a system user	Call forwarding via Forum Auto Attendant (with the **Connect to phone number** and **Connect to voicebox** function) or via Forum Voicemail (with the secretarial function)

Attributes

A call forwarding chain can contain any call forwarding types and call forwarding users. There is no limit to the number of successive call forwarding instances.

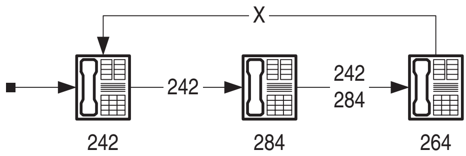
If the call forwarding destination is a system telephone, an incoming call is additionally indicated with the display `via...`. The caller list of a system telephone can also determine both the call number of the caller as well as the call number of the user doing the forwarding.

In the case of multiple forwarding, a setting in the user group of the call forwarding destination determines which of the “Via” call numbers is displayed. You can have either the last call forwarding user (default) displayed or the first call forwarding user in a chain displayed. The “Via” call number is, however, only displayed when the call number display is activated for the forwarding user.

Note: *When call forwarding to the Forum Voicemail voicebox programme, the “Via” call number is evaluated in order to determine the owner of a voicebox. The last call forwarding user is used no matter what the user group setting. When call forwarding using the secretarial function of the voicebox programme, the call number of the voicebox owner is also shown as the “Via” call number.*

Loop Detection

Loops can generally occur during a call forwarding chain, e. g. when the call forwarding destination refers back to the call forwarding source. This is why a forwarded call has a call forwarding history. When the next call forwarding destination is already included in the call forwarding history, a loop is detected and any further call forwarding is prevented. If no parallel call signalling takes place, e. g. by the **Indicate call forwarding after time parallel** setting in the user group, the call is cleared when there is a loop.



Loop Detection

A loop is also detected during call diversion. When you wish to divert an incoming call to a destination call number which is already part of the call forwarding chain, the display shows **NEG.** and call signalling is continued.

Note: *The call forwarding history cannot be transmitted via Q.SIG connections. Chain detection is also deactivated when forwarding via the voice portal programme Forum Auto Attendant.*

Virtual Call Numbers

A virtual call number is not assigned to any terminal. You always also enter an internal or an external destination call number directly when configuring a virtual call number. When the virtual call number is called this destination call number is signalled. This behaviour is handled as an immediate call forwarding and is thus the first call forwarding in a possible call forwarding chain.

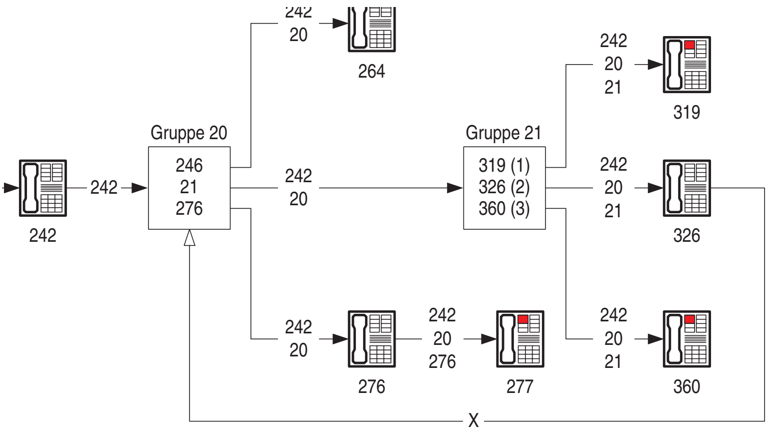
You can include a virtual call number e. g. in call distribution. Using the possibility of multiple call forwarding, you can also use a virtual call number as an exchange (“Operator”). Furthermore, a user with the **Call forwarding for other user** authorisation can configure additional call forwarding for a virtual call number also.

Hunt Groups

A hunt group is an internal call number which can reach multiple users. An incoming call is signalled to all users of a hunt group. A hunt group is configured in the **Configurator** in the **Telephony**: **Groups**: **Hunt groups** menu. Call signalling is independent of the **Type** setting in the hunt group:

- **Parallel** setting: all users of the hunt group are called simultaneously.
- **Linear**, **Cyclical** or **Statistical** setting: the users of a time-dependent hunt group are call successively. The user called first is determined by the respective setting.

The users of a hunt group can also configure their own respective call forwarding. For this, you need to activate the **Call forwarding of members possible** option for the desired hunt group on the **Telephony**: **Groups**: **Hunt Groups** page. The call forwarding history is forwarded to multiple users and continued separately respectively. When a call forwarding loop is detected for a user, no further call signalling takes place for this user. This user is not called in a parallel hunt group. This user is skipped in a time-dependent hunt group.



Forwarding hunt groups

Using a call forwarding chain, you can include a further hunt group as a user in a parallel hunt group.

Note: *Time-dependent hunt groups do not permit including any additional hunt groups.*

In addition, you can also configure call forwarding on busy (CFB) for hunt groups. This call forwarding can be used for an internal or external destination call number. In addition, you can enter a back-up destination for time-dependent hunt groups when they cannot be reached. Using the **Forward after time** setting, you can instead configure a call number and a specific time interval in the **Telephony**: **Groups**: **Hunt Groups** menu.

External Call Forwarding

You can also forward calls to external destinations without restriction. However, the call forwarding history cannot be continued with external destinations. When multiple users of a parallel hunt group configure call forwarding to external destinations, a single call to the hunt group can also occupy multiple external lines.

Special rules apply for call numbers which are displayed upon an external call forwarding destination:

- If the call is originally initiated by an internal user, the external call number of the internal user is transmitted.
- If the call is originally initiated by an external user, this user's external call number can be transmitted using the“CLIP no screening” feature. If “CLIP no screening” is not available, the external call number of the last forwarding user is transmitted.

Information on the Update

It is possible to convert the configuration of an earlier firmware version to the current version within the context of an update. This happens automatically when you load a saved configuration after an update. Please note the following points:

- The previous behaviour is preserved with the conversion. When restoring a former configuration, the **Call forwarding one level only possible** option is switched off automatically on the **Telephony**: **Settings** page. Additionally, the **Call forwarding of members possible** option is deactivated for all restored hunt groups.

- Restored call forwarding instances can behave differently as call forwarding chains of earlier firmware versions (before the 6.0 release) are not supported. If necessary, use the trace function for call forwarding (the **Diagnosis**: **Log Trace** menu, **CF tracking** option in the **Configurator**).
- The call forwarding interval, previously only centrally definable, is adopted for all users as the default value.
- The “Show hunting group no. as via” option for hunt groups has been omitted. Instead, use the **Display: Call forwarding via** option in the **User Manager**: **User groups** menu.
- The “Courtesy Service” function used to be for directly determining whether a user was currently busy or not and the corresponding announcement played. The **Announcer at Incoming call** is always played when it is configured with the firmware version (7.0 release). Then **Announcer at busy** is played if this announcement is configured.

Note: *Please note the explanations in chapter Update Notes starting on page 28 as well.*

Busy Keys

You can configure busy keys on a system telephone or on a SIP system phone. Busy keys show the busy status of another subscriber and can also be used for the selected dialling and pickup functions. Busy keys are configured on the **Telephony**: **Devices**: **System phones** page in the **Configurator**. Select the **Function** entry from the **Type** picklist when configuring a key on the system phone. Select the **BLF** entry from the **Type** picklist when on a SIP system phone.

In contrast to previous versions of the system software the type of busy key has been changed from **Trunk key** to **Function key** in Release 7. This means that busy keys can be programmed on any number of devices in release 7. A previously existing configuration of busy keys is automatically converted when the system software is updated.

Tip: A busy key for virtual call numbers with destination call number is not supported (see *Virtual Call Numbers* starting on page 171). It is possible, however, to configure a busy key for a virtual call number without destination call number. When the virtual call number is called, the call is signalled by the busy key. The call can be retrieved with the busy key.

Busy keys on the system network

Busy keys on the Forum 523/524 communications network are realised as a subscriber model. A busy key requests busy information to do so. The destination of a busy key keeps a dynamic list of terminals which have to be reported when there is a status change. This function mode is optimised for using busy keys on the system network (see *PBX Networking* starting on page 152). Please note the following points:

- Busy key information can be transferred via all connection types of a PBX system networking: Q.SIG-ISDN, Q.SIG-IP and SIP-Tie-Line. The **Protocol extension** option has to be activated in the bundle configuration for Q.SIG-ISDN.
- Busy keys only function between Forum 500 / 5000 communications systems with the release 7 system software.
- If there are multiple lines available in a bundle for PBX system networking, one of the lines is selected for busy-key signalling. If this line fails, it can take a few minutes until the dynamic subscriber list is updated.

Note: *The "call number busy" busy-key function is the only one supported for PBX system networking. The "device busy" busy-key function requires device ID transmission. This is not supported by PBX system networking.*

PIN Code Telephony

The users in a company normally use the existing terminals primarily for company-related communication. Now the users would like to also be able to make private calls in some cases. Private calls require:

- seizure of special external lines,
- changed and valid authorisations,
- recording separate charges and
- that destination call numbers not be saved for redialling.

This function can be realised using the “PIN Code Telephony” feature. To initiate a special call, the user on any terminal uses a special menu function or code procedure. After the user enters his/her own internal call number and associated user PIN, the desired attributes for the next call are activated.

When the call is finished, the previously active telephone configuration is restored. This external call number dialled is not recorded in a redial list.

Configuration

You can configure the “PIN Code Telephony” feature separately for each company. Configuration is done with the following steps:

1. Call the Forum 523/524 communications system **Configurator**. On the introductory page, change the **Level** option to **Expert**.
2. Open the **User Manager**: **User groups** menu page. Click on the **New** button. Enter a designation into the **Group name** field and select an existing user group as a template. Confirm with **Apply**.
3. Then click on the **Change** button. Select the desired **Company**. Activate the **PIN dial** option for the time groups intended. Configure either external authorisation or recording of **Connection data**. Confirm with **Apply**.

4. Open the **Telephony**: **Extended**: **Companies** menu page. Click on the headline of the company desired (default: “Company 1”).
5. Under the **PIN dial** header, select the desired **Route**. Select the user group created by the previous steps for the **User group** setting. Confirm with **Apply**.

You can also use the “Standard” user group or the “External trunk” standard route for the “PIN Code Telephony” feature. If you configure a special route, you must also enter a seizure code for this route (**Telephony**: **Trunks**: **Route** menu, setting: **Code**).

Activate the **PIN dial** user group option for all terminals which are to have the “PIN Code Telephony” feature available to them. This can also be the “Guests” user group. Furthermore, you can also use the “PIN Code Telephony” feature for users who have no terminal assigned to them (“virtual users”).

Implementation

You would like to use the “PIN Code Telephony” feature from any telephone. To do so, the telephone belonging to another user is switched to your personal user account (“identity change”). Carry out the following steps:

1. Call the main menu on a system telephone. Select the **6 Connections**: **7 PIN dialling** menu entry. Enter your call number and your user PIN. If the entry is correct, **<PIN dialling>** now appears. Now select the desired external call number including the seizure route for the “External Trunk” standard route.

   You can also initiate PIN dialling with a code procedure:
   # 4 6 (your no.) * (user PIN) # (external no.)

   The route configured for the “PIN Code Telephony” feature is used to make a call in that the seizure code for the standard route is also replaced by another seizure code if applicable. Furthermore, the authorisations are activated for the user group designated for the “PIN Code Telephony” feature.
2. Make your call. Please note that the call number identity being used is displayed to the external caller even when you are

calling from a different terminal. In addition, the call number identity being used is indicated as busy during the call. This is why the corresponding trunk key also lights up during this time on the respective system telephone.

The external call number dialled is not saved in any redial list: neither on the telephone used, nor one's own telephone.

Switch authorisation

The “Switch authorisation” feature enables a user to switch the user group of another terminal for a single call. The user switching can receive a charge notification at the end of the call.

A typical application is the guest telephone in a hotel: the concierge on duty activates external dialling for a guest when desired. When the call is finished, the concierge is informed of call duration and relevant charges with a brief information display.

Configuration

You can configure the “Switch authorisation” feature separately for each company. Configuration is done with the following steps:

1. Call the Forum 523/524 communications system **Configurator**. On the introductory page change the **Level** option to **Expert**.
2. Open the **User Manager**: **User groups** menu page. Click on the **New** button. Enter a designation into the **Group name** field (e. g. “Guest external seizure”) and select an existing group as a template. Confirm with **Apply**.
3. Then click on the **Change** button. Then select the desired **Company**. Configure the external authorisation and recording to **Connection data** as desired. Activate the **Immediate External line seizure** option as desired. Deactivate the **Switch authorisation** option to prevent authorisation from being activated for an additional call in an unauthorised manner. Confirm with **Apply**.

4. Click on the user group for the user who is to execute the “Switch authorisation” feature. Activate the **Switch authorisation** option. Confirm with **Apply**.
5. Open the **Telephony**: **Extended**: **Companies** menu page. Click on the header of the company desired (default: “Company 1”).
6. Under the **Switch authorisation** header, select the **User group** temporarily active due to switching (e. g. “Guest external seizure”). Activate the **Notify at end of call** option as desired. Confirm with **Apply**.

Devices to be switched (“guest telephones”) are generally configured in a user group without external authorisation. If desired, you can decline to assign these terminals to a user. In this case, the “Guests” user group is automatically assigned.

Implementation

A hotel guest would like to make an external call. You fulfil this wish with the following steps:

1. Call the main menu on your system terminal. Select the `6 Connections: 6 Switch auth.` option. Enter the internal call number of the terminal where a temporary authorisation switch is to take place. Confirm with the `OK` key. Select the `On` option and confirm with the `OK` key.

   If the terminal to be switched is currently in the call state, the authorisation switch only takes place once the call is finished.
2. The next outgoing call on the switched terminal then is made with the changed authorisations. If, e. g. the **Immediate External line seizure** option was activated to do so, the external dialling tone is audible and an external call can be made without entering a code.

   If the next outgoing call is not made within 60 seconds, the authorisation switch expires automatically.

At the end of the call – when configured in this way – you receive a brief message regarding the call duration and the relevant charges incurred.

Team Functions

Introduction

With the team functions you can manage your telephone communication tasks by assigning lines with separate call numbers to the keys of different terminals. The terminal users, or team members, can thus pick up one another's calls or telephone each other using the configured keys.

Team functions can only be configured on the Forum Phone 520 / 530 and Forum Phone 515 / 525 / 535 system telephones because only these have the required features.

Explanation of Keys

The team functions are programmed on the call keys of the Forum Phone 520 / 530 and Forum Phone 515 / 525 / 535 telephones. Depending on the terminal, different numbers of call keys are available:

Number of available call keys

Telephone	Number of keys
Forum Phone 520	Three keys with a display, five keys without a display
Forum Phone 530	Nine keys with a display
Forum Phone 530 with Forum Keypad 530	19 keys with a display, nine on the telephone itself and another 10 on a Forum Keypad 530
Forum Phone 515	One key with a display, five keys without a display
Forum Phone 525	Three keys with a display, five keys without a display

Number of available call keys

Telephone	Number of keys
Forum Phone 535	Nine keys with a display
Forum Phone 525 or Forum Phone 535 with an additional key extension Forum 500 Keypad Paper FP 525 / 535	36 additional keys without a display Up to three of these key extensions can be used with an Forum Phone 525/ Forum Phone 535.
Forum Phone 535 with an additional key extension Forum 500 Keypad Display FP 535	20 additional keys with a display Up to three of these key extensions can be used with an Forum Phone 535.

Note: *Only one function or call number can be programmed for each call key.*

The following keys can be used:

- **Trunk key**: Calls (for the programmed call number, e.g. 11) are signalled to this key, and you can make internal and external calls via this number. A trunk key can be programmed with a substitute function (with another team member acting as the substitute). Calls for you are then signalled to the terminal of another team member. A trunk key also provides functions for managing calls. For example, you can configure call protection if you do not want to be disturbed, or call diversion to another telephone.
- **Team key**: As with a trunk key, a team key can be used to receive or make calls. However, this key cannot be used to change the settings for managing calls; it is not possible, for example, to configure call diversion to another telephone. Calls made via a team key are signalled to all terminals with a trunk key that has been programmed with the same number. For example, the team key with the number 11 calls all trunk keys with the number 11.
- **Direct call key**: Only outgoing calls can be made with a direct call key; they are signalled to all terminals with the same number programmed to a trunk key. Calls via a direct call key are signalled to the destination terminal even if that terminal

has been programmed with a substitution function or call protection. If the destination terminal has been configured for call diversion, the direct call is not diverted.

Which Key is Suitable for Which Purpose?

- **Trunk keys** can be assigned call numbers for managing central communication tasks, for example, customer support. If the call numbers of the support department are assigned to trunk keys on all of its terminals, then all members of the support department can receive and manage calls and use the substitute function.
- **Team keys**, for example, can be used to create a project group within a department. Calls from customers of this group can then be answered by any team member who is not busy. The team members can call each other by the team keys.
- **Direct call keys**, for example, can be configured at a terminal in a conference room to call the secretary.

Tip: A **busy key** can be used to configure an enquiry station showing the status of the individual users. The enquiry station sees the status of the users and can put calls through by simply pressing the key (see *Busy Keys* starting on page 174).

Team Configuration

You can create teams and programme call keys in the **Configurator** of the Forum 523/524 (**Telephony: Groups** and **Telephony: Ports: Upn** menu).
Call key 1 is preset as a trunk key on all system telephones. This setting can be changed by the system administrator.

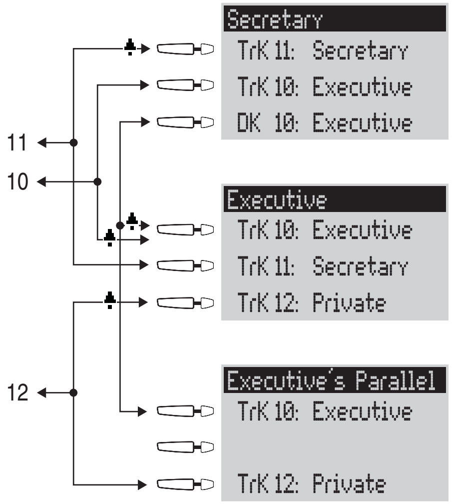
Examples of Use

The following examples illustrate the various uses of teams and team functions.

For information on the display texts and how to use the individual functions, refer to the chapter “Managing Calls in a Team” in the „Forum Phone 520, 530” or “Forum Phone 515, 525, 535” user guide.

Executive/Secretary Team

In this example, the executive/secretary team comprises two members: the executive and the secretary. The secretary has one Forum Phone 525 system telephone, and the executive has two, one of which is used as a parallel telephone in a sofa suite.



Example: executive/secretary team

Line Seizure

The secretary can be reached on the call number 11 (trunk key TrK 11: secretary’s office).

The executive can be reached on the call number 10 (trunk key TrK 10: executive’s office). He can also answer calls from his par-

allel telephone. In addition, a private line is configured for both of the executive’s telephones (trunk key TrK 12: private).

Call numbers 11 and 10 are both configured as a trunk key on the executive’s and the secretary’s terminal respectively. Thus the executive and the secretary can use either call number (for answering as well as making calls). Each can act as a substitute for the other.

The secretary’s terminal also has the executive’s call number configured as a direct call number (DK 10: executive’s office). The secretary can therefore reach the executive and put through calls even if the executive has programmed a substitute.

Line Busy Indication

If a line is busy, e.g. TrK 11 secretary’s office, the other terminal will indicate this. The executive’s private calls via TrK 12 are not indicated on the secretary’s terminal since no appropriate trunk key is configured on the latter’s telephone.

Call Signalling

In this configuration example, calls to one’s own call number are signalled acoustically on the following telephones:

- Call number 11 on the secretary’s telephone
- Call numbers 10 and 12 on the executive’s telephone.

Calls for the other team member’s call number are indicated by an optical signal on one’s own telephone (flashing trunk key LED).

The parallel telephone will indicate calls only by an optical signal.

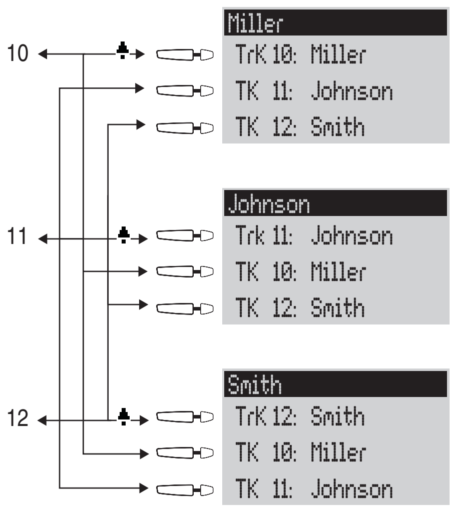
Time-delayed acoustic signalling can be configured for TrK 10 on the secretary’s telephone. If the executive, for example, does not answer a call within 10 seconds, the secretary’s telephone will start to ring.

If the executive activates a substitute function with the secretary as the substitute, calls for call number 10 will be indicated on the executive's telephone by an optical signal only, but signalled acoustically on the secretary's telephone. The secretary can also activate a substitute function. Calls for call number 11 are then signalled acoustically on the executive's telephone, and indicated by an optical signal on the parallel telephone and the secretary's telephone.

Three-member Team

The three-member team described here is an example of a team configuration within a project group, e.g. export sales.

Each team member has one Forum Phone 525 system telephone with all call keys programmed as trunk and team keys.



Example: three-member team

Line Seizure

Each team member's call number, e.g. call number 10 for Miller, is programmed as a trunk key on his telephone.

On the other telephones in the team, this call number is programmed as a team key (e.g. TK 10 on Johnson's and Smith's tel-

ephones). The team members can thus see which number a call is for and can answer it by pressing the appropriate team key.

The team members can call each other via the team keys. For example, Miller can call number 12 by pressing TK 12; the call is then signalled to Smith’s telephone on TrK 12.

Line Busy Indication

If a line is busy, e.g. TrK 11 Johnson, the team keys 11 on Miller’s and Smith’s telephones will indicate this.

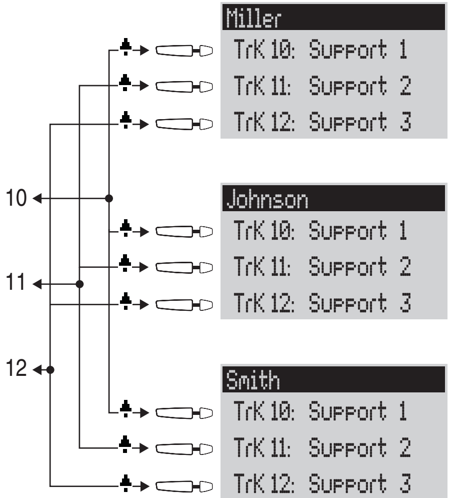
Call Signalling

In this example, calls via the trunk keys are signalled acoustically. Calls via the team keys are indicated by a visual signal (the team key LED flashes).

Unified Team

The unified team described here is an example of a team configuration within a department in which calls are to be managed quickly (e.g. support department).

Each team member has one Forum Phone 525 system telephone with all call keys programmed as trunk keys.



Example: unified team

Line Seizure

Call numbers 10, 11 and 12 are programmed as trunk keys on each team member's telephone (TrK 10 to TrK 12.

All team members can use these numbers for answering as well as making calls.

Tip: In this team configuration it is useful to programme one of the function keys on each telephone with the "Hold" function. A call, e.g. for TrK 11, can then be put on hold by pressing the function key. If another team member then presses trunk key TrK 11 on his telephone, he can accept the call. For further information on function keys, refer to the „Forum Phone 520, 530" or "Forum Phone 515, 525, 535" user guide.

Line Busy Indication

If a line is busy, e.g. TrK 11 Johnson, the trunk keys on the other team telephones will indicate this.

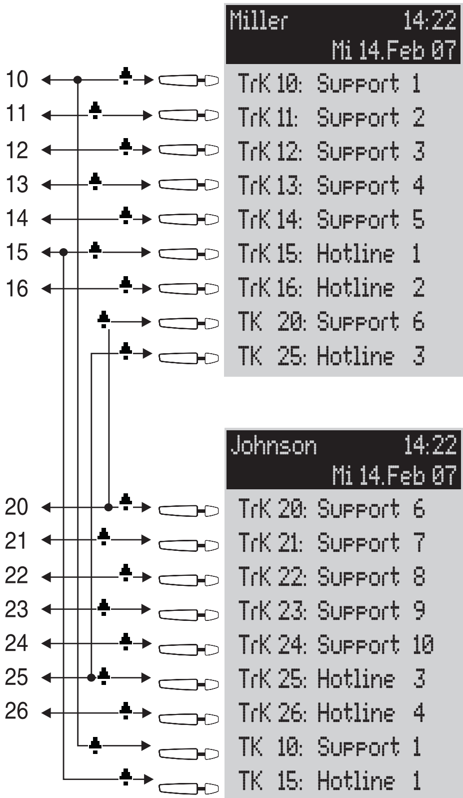
Call Signalling

In this example, calls via all trunk keys are signalled acoustically.

Toggle Team

The toggle team described here illustrates how a large number of call numbers can be managed efficiently with the help of team functions.

Each team member has one Forum Phone 535 system telephone with all call keys programmed as trunk and team keys.



Example: toggle team

Line Seizure

Each team member is assigned seven call numbers, each programmed as a trunk key (TrK 10 to TrK 16 and TrK 20 to TrK 26). For each member, these trunk keys are programmed either as support numbers or hotline numbers.

The first support number and the first hotline number of each team member is programmed as a team key on the other member's telephone, e.g. TrK 10 and TrK 15 on Miller's telephone as TK 10 and TK 15 on Johnson's telephone. The assumption here is that most calls will go to the respective first call numbers, and team members can thus help each other out by answering one another's calls.

On each telephone it is possible to toggle between the calls on individual lines, e.g. TrK 10 and TrK 11, by pressing the appropriate key (toggling).

Every call on a trunk key can be transferred to any other party by means of the R key. For more information, refer to the chapter entitled "Consultation, Toggling, Transfer and Conference" in the „Forum Phone 520, 530" or "Forum Phone 515, 525, 535" user guide.

Line Busy Indication

If a line is busy, e.g. TrK 10 on Miller's telephone, the appropriate team key will indicate this, e.g. TK 10 on Johnson's telephone.

Call Signalling

In this example, calls via trunk keys are signalled acoustically. Calls via team keys are indicated by a visual signal (the team key LED flashes).

Call Queue

Introduction

A queue can be activated for the telephone numbers of any type of telephone, i.e. for system, analogue, ISDN and DECT telephones. If a call number with a queue is busy, calls to this number enter the queue. The caller first hears an announcement (if function “Announcer at busy” is configured) and then a dial tone or Music on Hold, if the **Music on hold upon call transfer** setting is activated.

Calls which remain in the queue for too long are cleared from the queue. The caller then gets a busy tone. If all the positions in the queue are taken then any further calls also hear the busy tone. If the **Courtesy Service** setting is activated, calls will not be cleared from the queue!

The time until an external call is cleared from a queue is defined by the network operator. In the most European countries this is usually three minutes.

If more than one telephone number (e.g. trunk or team keys) has been configured for a telephone, separate queues are used for each number.

On the Forum Phone 535 or Forum Phone 530 system telephone, additional calls are signalled by a brief tone in the loudspeaker and in the display. If calls are in the queue, a number at the beginning of the second line of the display on the Forum Phone 535 / Forum Phone 530indicates how full the queue is. If more than one telephone number with a queue is configured on the telephone, the total number of entries are displayed.

Calls in a queue are handled by the Forum 523/524 in the following order of priority: instant connection, door calls, automatic recalls, VIP calls, then other internal and external calls. Sensor calls thus have priority over other calls, for example. Calls of the same priority level are switched in the order of their arrival.

The system administrator sets the number of calls that can be placed in a queue individually for each user group. The value can lie between “0” and “99”. The “0” value deactivates the ”Call queue” function for a user group. When the maximum number of calls in the queue is reached, further callers hear a busy tone.

Note: *As calling fax machines often operate with the “voice” service indicator (e.g. on analogue ports), you should assign ports for fax machines on the Forum 523/524 to a user group **without** a queue.*

Queues can be combined with the “call forwarding,” “pickup” and “hunt group” functions, for example, in order to configure an enquiry station for an operator.

Activation of Queues

Queues can be activated on a per user group basis. On delivery the default set, for all preset groups, is **off**.

When using queues, it often makes sense to activate call waiting protection. For this purpose, **Call waiting protection** authorisation must be allocated to the user group, and call waiting protection must be activated on the terminal.

Furthermore it is sensible to combine queues with the **Announcer at busy** function. When a caller calls a subscriber who is busy then they will hear a “central welcoming text”, for example, “Here is company XYZ. You will be immediately connected”. The function **Announcer at busy** can be set in the **Telephony: Call Distribution: Incoming** or **Telephony: Call Distribution: Incoming DDI** menu. Central welcoming texts can be recorded using the programme package Forum Voicemail.

You should configure a new user group (e.g. “Operators”) and activate the authorisations **Call queue**, **Call waiting protection** and, if necessary, **Call forwarding**. If users belong to this group, a queue will be activated automatically for all telephone numbers assigned to them.

Call Forwarding

Forwarded calls of the forwarding type “Immediately” and “On busy” have priority over queues. The queue of the forwarding telephone is not used for forwarding calls in this manner.

During the configuration of this type of call forwarding, the contents of the queue are **not** transferred to the target terminal. If there are still calls in the queue when the call forwarding function is activated, these calls can only be accepted on the source terminal.

If a call is to be forwarded “After delay”, it enters the queue. If the call has not been answered before the delay period expires, it will be forwarded to the target terminal and can then be answered there.

Pickup

The functions “Pickup from group” and “Pickup selective” can be used together with queues. A user who accepts a call using “Pickup from group” or “Pickup selective” picks up the next call from the queue.

Hunt Groups

Hunt groups of the “parallel” type are usually used together with queues, with the queues of each telephone in the group being synchronised to each other. When a call to the number of the hunt group arrives, the call enters all parallel queues. If a call from one of the queues is answered, it is removed from all other parallel queues.

Examples of Use

Enquiry Station for an Operator with Two System Telephones

The operator switches all incoming calls and can either work on the Forum Phone 535 or the mobile terminal, the Forum Free 566 / 576 / 586.

Configuration

- Configure the system access or access for multiple terminals under **Telephony: Ports: S$_0$**.
- Configure the Forum Phone 535 and e.g. a base station (RFP) under **Telephony: Devices**.
- Configure a trunk key for the Forum Phone 535 under **Telephony: Devices: System telephones**.
- Configure the Forum Free 566 / 576 / 586under **Telephony: Devices: DECT phones** and assign the Forum Free its own telephone number. Check in the Forum Free.
- Under **Telephony: Call Distribution: Incoming** or **Telephony: Call Distribution: Incoming DDI** route all incoming calls to the number of the Forum Phone 535trunk key.
- In the **Configurator**, create a new group called "Operators" under **User Manager: User groups**. Activate the **Call queue**, **Call waiting protection** and **Call forwarding** authorisations for this group and set the **Dial in (outgoing): External** option appropriately.
- Create a user called "Operator 1" under **User Manager: User**. Assign this user to the "Operators" user group. Assign the telephone numbers of the Forum Phone 535 trunk key and the number of the mobile Forum Free 566 / 576 / 586 to this user.
- Activate `Call wait. prot.` (call waiting protection) on both terminals in the `Protection` menu.
- Configure a feature key on the Forum Phone 535 which activates / deactivates a "call forwarding immediately" to the

telephone number of the mobile Forum Free 566 / 576 / 586 (in the menu `Call diversion: Divert phone: Immediately`).

Use

Incoming calls are routed to the Forum Phone 535 manned by the operator, who then puts the calls through. A queue is used so that callers do not get a busy signal. The display on the Forum Phone 535 indicates how many calls there are in the queue.

If the operator wants to leave the workstation and take along the enquiry station, call forwarding to the Forum Free 566 / 576 / 586 is activated by pressing a feature key. Calls which are in the Forum Phone 535 queue must still be answered on this telephone. New calls are signalled on the mobile Forum Free or enter its queue, allowing the Forum Free to be used as a mobile enquiry station.

On returning to the workstation, the operator deactivates call forwarding by pressing a feature key. Calls which are already in the queue are switched on the mobile Forum Free 566 / 576 / 586. New calls are signalled on the Forum Phone 535 or enter its queue.

Group of Three Enquiry Stations

The enquiry stations switch all incoming calls. Incoming calls are administered in queues. Depending on the number of arriving calls, one to three enquiry stations in this group are manned. The enquiry stations are each equipped with a Forum Phone 535.

Configuration

- Configure the multi-terminal access or the system access under **Telephony: Ports: S$_0$**.
- Configure the three Forum Phone 535 telephones under **Telephony: Devices**.
- Configure a trunk key with its own telephone number for each of the Forum Phone 535 telephones under **Telephony: Devices: System telephones**.

- Configure a hunt group of the **parallel** type under **Telephony: Groups: Hunt Group**, and include the three telephone numbers of the trunk keys in this hunt group.
- Under **Telephony: Call Distribution: Incoming** or **Telephony: Call Distribution: Incoming DDI** route all incoming calls to the number of the hunt group.
- In the **Configurator**, create a new group called “Operators” under **User Manager: User groups**. Activate the **Call queue** and **Call waiting protection** authorisations for this group.
- In the **User Manager**, configure a user for each of the three operators and assign these settings to the user group called “Operators”. Allocate each user the telephone number of the trunk key of their system telephone.
- Activate `Call wait. prot.` (call waiting protection) on all three terminals in the `Protection` menu.
- Programme a feature key with the function “Sign on / sign off from hunt group” on the three system telephones (in the menu `Calls: Hunt group`).

Use

Incoming calls are signalled in parallel to all signed-on enquiry stations. If the enquiry stations are busy, the incoming call joins the queue on each of the terminals in the hunt group. If one of the enquiry stations accepts a call from the queue, the call is removed from the queues of all the other enquiry stations. The display at each enquiry station (Forum Phone 535) indicates how full the queue is.

If attendants leave the station, they sign off from the hunt group by means of a feature key. In contrast to Example 1, further calls do not have to be processed after the sign-off, as the calls are also registered in the queues of the other signed-on enquiry stations.

Note: *The last enquiry station remaining in the hunt group should not sign off, so that incoming calls can always be signalled to at least one station.*

Multi-Company Variant

Communications systems are frequently shared by several companies. These companies want to jointly use the existing infrastructure (e.g. the existing lines and features of the system), while at the same time they wish to organise and pay for their communication completely independently of one another.

This "multi-company variant" can be implemented using the Forum 523/524 within a shared office, for example.

In the multi-company variant, the companies are essentially completely independent of one another. This allows them to have their own trunk lines, which is useful for billing purposes. The Forum 523/524 hardware and software are used equally by all the companies, however. It is possible to configure the Forum 523/524 for each company and define the extent to which the features of the system may be used.

In brief, the features of the multi-company variant are as follows:

- Up to 5 companies can be configured at the same time.
- Every user of the Forum 523/524 is assigned to a company.
- Each available bundle (trunk group) or SIP trunk is uniquely assigned to a company so that incoming external calls can be transferred to the correct internal subscriber.
- For each company, every route can have its own code. For example, it is possible to activate different routes with the code "0" for different companies. This enables separate charging for outgoing external calls, for example.
- An individual exchange ("operator") can be set up for each company.
- Each company can maintain the communication data of its business partners in its own company telephone book.
- The charges can be billed individually for each company.

Configuring the Multi-Company Variant

The multi-company variant can be commissioned and configured by the system administrator of the Forum 523/524 without any major effort. In the multi-company variant, the communications system behaves in exactly the same way as the single-company variant. This is particularly of interest to users who want to expand their own system and at the same time operate it in a group.

The process in brief:

1. The feature must be activated (see *Activating the Multi-Company Variant* starting on page 198).
2. The required companies must be set up (see *Configuring and Managing Companies* starting on page 198).
3. The users of the Forum 523/524 are assigned to the individual companies (see *Assigning Users* starting on page 199).
4. In order that the Forum 523/524 can transfer incoming calls to the corresponding company (or its staff) correctly, the existing bundles must be uniquely assigned to the companies (see *Assigning a Bundle/SIP Trunk* starting on page 199).
5. In the case of outgoing external calls, the lines via which the members of a company can make a call must be defined (see *Allocating Routing Codes* starting on page 200).
6. An exchange must be set up for each company so that the Forum 523/524 can correctly process statuses in which a call should be routed to the exchange (see *Configuring the Company Exchange* starting on page 200).

Activating the Multi-Company Variant

To be able to configure several companies in the Forum 523/524, the “Multi-company variant” feature must first be activated. This is done in the **Configurator** on the Web console in the **System: Common** menu. Activate the **Multi company** option here.

Only when this option has been activated are the fields required to configure the multi-company variant available in the other menus of the Web console, for example in the **User Manager: User groups** menu or in the **Telephony: Trunks** menu.

Configuring and Managing Companies

Up to 5 companies can be configured in the Forum 523/524. By default, one company with the name “Company 1” is predefined. All configuration settings, e.g. in the user groups or in the bundle configuration, apply to this predefined default company if not other company has been selected.

Companies are set up and managed in the **Telephony: Extended: Companies** menu:

- A new company is created in this menu using the command **New**. Each company can be given a name up to 20 characters long. This name is then displayed in all configuration dialogue boxes in which company-specific settings can be defined.
- In this menu a company can be deleted again using the command **Delete**. If a company is deleted which is still used at other places (in the user groups, for example), the respective configuration is changed to the default company.
- The name of the default company can be changed, but the default company itself cannot be deleted.

Assigning Users

For each user you must define the company to which they belong. This assignment determines, for example, which company telephone book the user has access to and which company-specific configuration data apply to them.

As the Forum 523/524 manages users in groups, the assignment “user > company” is also established this way. The company to which each user group belongs must be defined for each group. A user group can only belong to one company, i.e. not to several. However, a company can have several user groups. It is therefore possible, in the same way as in the entire system, to allocate a range of authorisation rights for the use and configuration of features for each company.

When setting up a new **User group** (in the **User Manager** menu), you will find that the default company is predefined; another company can be assigned as long as no other companies have been set up.

Assigning a Bundle/SIP Trunk

Connections of the same type and in the same direction are arranged in a bundle (e.g. S_0 multi-terminal connections). To be able to correctly transfer incoming calls to the members of the configured companies (the users) via the lines of a certain bundle of the Forum 523/524, each of the available bundles must be assigned to one of the companies. This is necessary to be able to transfer incoming external calls to the correct company exchange in cases where the called internal subscriber cannot be reached, for example. SIP trunks can also be assigned to a company.

The assignment of bundles to companies is done in the **Telephony: Trunks: Trunk group** menu. The assignment of SIP accounts to companies is done in the **Telephony: Trunks: SIP trunks** menu.

Note: *A bundle configured as a **System access** can however also be used jointly by multiple companies. Available direct dialling-in numbers of the system access can be assigned to the internal company call numbers via the **Telephony**: **Call Distribution**: **Incoming DDI** menu.*

For outgoing external calls which users set up via the lines of their company’s bundle/SIP trunk, the assignment of the bundle to the company is irrelevant: the charges are assigned according to the “source” principle.

Charges are billed to the company to which the user belongs who set up the connection. The Forum 523/524 recognises this on the basis of the assignment between user groups and companies and on the basis of the routing code with which a line of the bundle/SIP trunk was seized. For more information, please see the following section.

Allocating Routing Codes

Routes are used for automatic and selective seizure of bundles or connections for external calls. It is possible to seize a route by predialling a code.

In the **Telephony: Trunks: Route** menu, you can define which company can seize each route. An individual **code** for the seizure is allocated per route for each company. The Forum 523/524 ensures that during configuration no seizure code is allocated twice (for two different routes) for each company. If during configuration of a route no code is allocated for one of the configured companies, the route concerned cannot be seized by the members (user groups) of this company.

Configuring the Company Exchange

An internal telephone number must be set up for each company which represents the exchange, i.e. “the operator”. The calls to specific extensions arriving at the exchange are routed to this number, for example, as are all external calls where the called subscriber (a user who belongs to this company) cannot be reached, as in the case of a timeout.

A company exchange is set up in the **Telephony: Operator** menu. In this menu, you can specify an internal telephone number for each company and time group which then represents the exchange for this company.

Working with the Multi-Company Variant

All the features of the Forum 523/524 which the users may already be familiar with from the single-company variant are available in the multi-company variant. These features can be used to the same extent and can be used in exactly the same way.

The following section describes the features additionally available to the users in the multi-company variant.

Company Telephone Book

An individual company telephone book can be created for each company. In addition to this, "personal" and "central" telephone books exist:

- A personal telephone book is available for each user.
- The central telephone book can be used across the companies by all users of the Forum 523/524.

The company telephone book is a central telephone book for the whole company. It is only available to the users/user groups who are assigned to this company. You can also define whether the members of each user group may edit the company telephone book or not.

The company telephone book is treated exactly the same way on the system terminals as the other types of telephone books. This means that the entries listed in the personal, central and company telephone books are displayed on the system phones at the same time.

Users can also use the telephone book of their company with the **Forum CTI** and **Phone Book** Web applications, assuming they are authorised to use these applications.

In addition, it is also possible to assign a user group with the authorisation to edit foreign company telephone books. This authorisation is useful if members of this group - e.g. the "Admin-

istrators" - service the entire system. Foreign telephone books can only be edited in the **Configurator** in the **Phone Book** menu.

The number of entries in a company telephone book is unrestricted. The Forum 523/524 can manage up to 2,000 entries in *all* telephone books (in the central, personal and company telephone books).

Making Calls Between Companies

All users of the Forum 523/524 can make internal calls to one another, irrespective of which company they belong to. Calls between users from the different companies are therefore not subject to any restrictions.

Billing Charges per Company

In the **Costs** Web application you can output the charges for each company.

Users who are authorised to use this application can view the charges for each company.

Configuring the PC Software

Further possibilities of use can be implemented on a workstation computer with the Windows operating system by installing drivers and programmes.You can find the installation programmes required for this on the product CD that comes with the Forum 523/524.

Proceed as follows to install extra software:

1. Log on under Windows as the administrator.
2. Insert the product CD.

   If your PC is suitably configured, the CD will start automatically. Otherwise select **Run** from the Start menu. Click on the **Browse** button to look for the programme "cd_start.exe" on the CD. Confirm this with **Open** and **OK**.
3. Select the required option from the start interface. Follow the programme instructions.

Further instructions for various options that are available are given below.

Using the Systray Display

You can configure a systray display for the Forum 523/524 to appear in the information area of the Start bar of a workstation computer. This systray display constantly shows you whether a WAN, a RAS or a Branch connection via ISDN is active. It is also possible to display the current occupancy of the trunk lines.

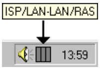
Requirements

To use the systray display, you must first install TAPI; see *Setting up TAPI Interface* starting on page 209.

Please note: *The systray display requires a current version of TAPI. If you are using TAPI from an earlier version of the Forum 523/524, you must first install the newer version from the product CD.*

Installing the Systray Display

1. Call up the start mask of the product CD (see *Configuring the PC Software* on page 203).
2. From the start mask, select **Software**: **Install Systray**. Follow the programme instructions.
3. Start the programme with **Start**: **Run** and the configuration dialogue is displayed. Select one of the entries displayed under **Existing PBXs**. Enter your user name and password in the boxes under **Log-on**.
4. If you activate the **Autostart** check box, you will see the systray display even after restarting your workstation computer.
5. Confirm the entries in the configuration dialogue with **OK** and the systray display logs on for the Forum 523/524.
6. Right-click on the systray display in the Start bar. Select **Configuration** to call up the configuration dialogue. Select **Network Connections** or **Trunk Lines** to produce a status dialogue.



Browser for Forum CTI and Forum Hotel

You can simplify the daily use of the **Forum CTI** and **Forum Hotel** Web applications using the Web browser especially adapted for the Forum 523/524. Each time the workstation computer is restarted, this browser programme can automatically start and log you in. This means that the applications are always operational and can be accessed using the icon in the information area of the task bar.

Installing the Browser

1. Call up the start mask from the product CD (see *Configuring the PC Software* on page 203).
2. Select **Software**: **Browser for Forum CTI** or **Software**: **Browser for Forum Hotel** from the start mask.
3. Follow the programme instructions.

After installing the browser, there is a new menu item in the Windows start menu under **Programs**: **Forum CTI Browser** respectively **Programs**: **Hotel Starter**.

Further information can be found in the online help of the browser programme. To view this, click the top left corner in the **Forum CTI**-browser's programme window on the system menu symbol or on the symbol in the information area of the task bar. Select the **Readme** command. You will find the **Forum Hotel**'s readme in the installation directory of this browser programme.

Note: *Both browser programmes can be used simultaneously.*

Synchronising the PC Clock

With the network service SNTP (simple network time protocol) it is possible to synchronise the internal clock of a workstation computer with the time of the Forum 523/524.

Requirements

You must enter the time zone so that the Forum 523/524 can calculate the time of the internal clock back to the GMT (Greenwich Mean Time) required for SNTP:

1. Go to the **Configurator, System: Common** menu. Click on **Change**.
2. Enter the **Time zone** for which the time of the Forum 523/524 applies and whether summer time is allowed for. Confirm this with **Apply**.

Configuring SNTP

For various operating systems, you can use one of the numerous SNTP programmes offered for downloading on the Internet. Configure the Forum 523/524 as an SNTP server for such programmes.

Please note: *In a Windows domain network, the PDC server (primary domain controller) should automatically assume the function of the timer.*

SNTP with Windows XP

Here you configure the SNTP server by double-clicking on the time in the Start bar. Enter the Forum 523/524 as the **Server** in the **Internet time** tab.

Application Interfaces

The Forum 523/524 communications system supports a series of application interfaces which are used by external systems to access functions of the communications system. Communication between the external system and the Forum 523/524 communications system is principally transported via the LAN connection or using the IP network protocol. The respective functions differ depending on the application interface. The properties, function and requirements of the various application interfaces are explained in the following sections.

CSTA interface

Using the interface for “Computer Supported Telecommunications Applications” (CSTA, phase 3 based on ECMA-269 and ECMA-285 (BER) /ECMA-323 (XML), phase 2 based on ECMA-217 and ECMA-218) external programs and applications can access services of the Forum 523/524 communications system via network connection. The CSTA interface provides an effective and universal connection option for Computer Telephony Integration (CTI) applications, for example:

- CTI functions for corded and cordless system terminals as well as SIP terminals
- Display paging and alarm messages on terminals
- Remote activation using terminals, e. g. function and status keys for device or building control.
- interactive menu control using telephony terminals possible
- Change administration information such as user groups, user names and display languages
- Calculating charges, wake-up calls, transparent keypad data with codes, transmitting charges, time group query, message calls and CTI for virtual subscribers, message calls and more.

The CSTA interface based on phase 2 with a limited range of functions available without licence, range of functions comparable with the TAPI basic functions. You can use the following CTI applications without a CSTA licence:

- Forum Count 500(0): This charging software can query and evaluate charge information via the CSTA interface. There is a CSTA to V.24 converter ("Count4CSTA") available which is a standard part of the charging software for doing so.
- SNMP Agent This Windows service program facilitates integrating the Forum 523/524 communications system into SNMP-based network management. You do not require a CSTA user log-on to operate this service programme.

To use the full CSTA interface performance, you have to activate the extended functions (CSTA phase 3 BER / XML) with a licence key. Activate the licence key in the **Configurator** on the **System**: **Licences** page.

CSTA interface access is also protected by the password a user account. You can configure a new entry in the **Configurator** on the **User Manager**: **User** page. When creating the user entry, you have to select the "Administrators" user group.

Authorisations can be assigned via user groups for using CSTA functions on terminals. To do so, configure the options under the **CSTA** header on the **User Manager**: **User groups** page. Please consult the notes in the online help regarding this page in the **Configurator**.

You can also develop your own CTI applications or adapt existing CTI applications for the CSTA interface. To do so, you need the CD-ROM, the developer documentation, library files and sample applications, available separately. Contact your Belgacom agent if you need this CD-ROM.

Setting up TAPI Interface

With a TAPI (**T**elephony **A**pplication **P**rogramming **I**nterface) you can operate a CTI application (**C**omputer **T**elephony **I**ntegration). Here, the CTI application uses the services of the Forum 523/524 with the help of the TAPI driver installed on a Windows PC.

Many telephony functions, such as enquiry, toggling, three-party conference, pick-up, call protection and call forwarding can be controlled using appropriate TAPI-compatible software.

Requirements

You require an active IP network connection between the PC and the communications system. CTI functions can be used only in conjunction with system terminals.

You must therefore have configured at least one user for a system terminal. In addition, you require a TAPI 2.1-compatible CTI application, for example the **Phone Dialer** included in the Windows operating system.

Note: *You can also install the TAPI driver under Windows 7 and under Windows Server 2008 R2 (x64). For Windows Server 2008 R2 (x64) an additional runtime library is required. Download and install the "Microsoft Visual C++ 2008 SP1 Redistributable Package (x64)" from the Microsoft web site under the following URL: http://www.microsoft.com/downloads/details.aspx?FamilyID=ba9257ca-337f-4b40-8c14-157cfdffee4e*

Installing the TAPI Driver

1. Call up the start mask from the product CD (see *Configuring the PC Software* on page 203).
2. Select **Software**: **Install TAPI Service Provider** from the start mask.
3. Follow the programme instructions.

Configuring the TAPI Connection

Note: *Under Windows you should log on as the user for whom you want to configure the TAPI connection.*

1. In the Start menu, select **Settings**: **Control Panel**. Select the **Printers and other Hardware** (with Vista/Windows 7: **Hardware and Sound**) category. Double-click on the **Phone and Modem Options** icon.
2. Change to the **Advanced** tab.
3. From the list of installed driver software, select **Forum 500 Service Provider** and click on **Configure**.
4. In the following dialogue you will find a list with the configured connections for the user who is currently logged on. Click on **New**.
5. In the following dialogue you provide information for the new connection. In the **Connection name** box you can enter a descriptive name for the connection. In the **CTI server** box you must enter the DNS name or the IP address of the Forum 523/524. Using the **[...]** button you can search for this in the LAN. In the boxes **Username** and **Password** you enter the user data of one of the users configured on the Forum 523/524. This user must be allocated a system terminal. Confirm your entry with **OK**.
6. The new connection is now configured. Close the opened dialogues with **OK** and **Close**.

Testing the TAPI Function

1. In the Start menu, select **Programs**: **Accessories**: **Communication** and then start the programme called **Phone Dialer**.

   Under Windows XP and Vista, the **Phone Dialer** is started indirect by using the dialling function of the **Address book** (can be found in the start menu under **Programs**: **Accessories**). A manual start of the programme file “Dialer.exe” in the “C:\Program Files\Windows NT” (XP) or “C:\WIndows\System32” (Vista) folder is possible also.
2. In the **Tools** menu, select the item **Connect using...** to select the system terminal that is to use the CTI application. Under Windows XP you select the item **Options** from the **Edit** menu. In the **Lines** tab you then select the system terminal from the **Phone calls** list.

3. Enter a telephone number in the **Number** box and confirm with **Dial**. Under Windows XP you first click on the **Dial** icon and in the subsequent dialogue activate **Phone call**.
4. The number you entered is displayed on the selected system terminal. Lift the receiver to start dialling.

Note: *This note is not relevant to Windows XP or Vista. If the "Phone Dialer" programme is not installed, you will have to install it. To do this, you open the* ***Control Panel*** *and click on* ***Software****. In the* ***Windows Setup*** *tab you activate the* ***Connections*** *component.*

LDAP Interface

Directory information can be transferred using the LDAP protocol. A central service (LDAP server) manages a comprehensive, hierarchically organised directory which can be queried by an LDAP client via LDAP. The Forum 523/524 communications system supports two different LDAP applications:

- Forum 523/524 communications system as LDAP client: an external directory can be queried via LDAP, for example, in order to display the information thus gathered when there is a call.
- Forum 523/524 communications system as LDAP server: the telephone book managed by the Forum 523/524 communications system can be accessed via LDAP.

Address Queries using LDAP

You can search the data of the central telephone book of the Forum 523/524 from a workstation computer in the LAN using LDAP (Lightweight Directory Access Protocol). When configuring an LDAP-enabled programme, specify the IP address of the Forum 523/524 as the address of the LDAP server.

LDAP with Outlook Express / Windows Mail

You can configure and operate the LDAP directory service with “Outlook Express” or “Windows Mail”, an e-mail programme, as follows:

1. Call up the **Accounts** command in the **Tools** menu.

   The **Internet Accounts** dialogue box will then open.

2. Click on **Add**. Select the **Directory Service** command from the pop-up menu.

3. Under **Internet directory (LDAP) server, enter the address of the** Forum 523/524. It is not necessary to enter log-in data to query the central telephone book. You have to enter the log-in data for the account on the Forum 523/524 communications system here if you wish to also query entries from the central telephone book or from the company telephone book additionally. Click twice on **Next**. Then click on **Finish**.

4. Check the function. In the **Edit** menu, call up the **Find**: **People** command.

   The **Find**: **People** dialogue box will then open.

5. In the **Look in** list, select the entry with the Forum 523/524 address. Enter a user in the **Name** input field, Administrator for example. Then click on Find now.

   The list of entries found should now display the address from the central telephone book.

Querying External LDAP Server

The Forum 523/524 communications system can also access external address directories. This means that alongside the internal telephone book entries, address information from external sources is also available. This information is used, for example, in a name search or for displaying a name for an incoming call.

The external address directory is queried by the Forum 523/524 communications system via the internet Lightweight Directory Access Protocol (LDAP). To do so, you require a server on the company network which makes address information available via

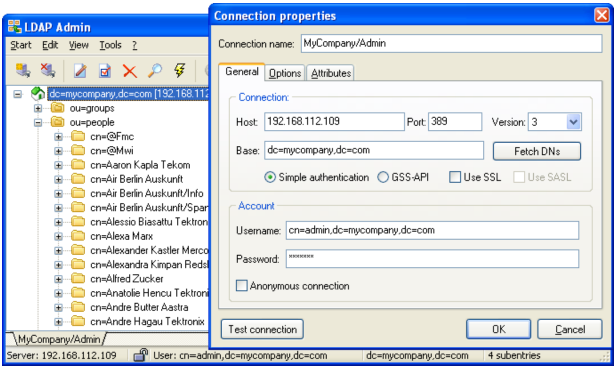
LDAP version 2 or via LDAP version 3. The LDAP query requires either an anonymous LDAP directory access without a password or a LDAP directory access with user name and password.

Note: *SSL encryption of LDAP communication is not supported.*

You can configure a separate LDAP directory access for each company configured (see *Multi-Company Variant* starting on page 196):

1. Open the **Telephony**: **Extended**: **Companies** page in the web configurator. The list of companies configured is displayed. If you do not use the multiple company variant, "company 1" is the only entry.
2. Open the configuration page for the company desired. Activate the **Status** option under the **LDAP Server** header. Enter the IP address or DNS name of the LDAP server into the **Server** entry field. The online help of the Forum 523/524 communications system provides detailed descriptions of the other entry fields.
3. Use **Apply** to confirm your settings.

An LDAP server can be configured to saved various types of information which are available in a tree-like data structure. Alongside querying address book entries, which should be saved in the "InetOrgPerson" schema, many companies also use a LDAP server for centrally organised management of user accounts, for instance. The following screenshot provides an example:



To configure the LDAP query you need to find out from the LDAP server administrator what part of the subtree contains the desired address information. The subtree is designated with a technical address (DN, “Distinguished Name”). This technical address (in the above example “ou=people,dc=mycompany,dc=com”) can be defined in the web configurator on the **Telephony**: **Extended**: **Companies** page with the **BaseDN** setting. If the LDAP server does not allow an anonymous query, configure the settings **User name** and **Password** for a LDAP user account with read authorisation.

Usage of address information queried via LDAP is activated as the default setting for all user groups. You can deactivate this function for individual user groups. To do so, deactivate the **Use LDAP** option in the web configurator on the **User Manager**: **User groups** page under the **Phone book** header.

After configuration, the information queried via LDAP is available on authorised system telephone, DECT system terminals and SIP system telephones as well as in the applications “ Forum 500 CTI” and “Forum CTI Touch”. Information queried via LDAP is considered for telephone book searches and for displaying names during incoming calls in addition to the information from the internal telephone book. Please note the following:

- When you have configured the LDAP query for a company, this changes the telephone book search for the DECT system terminals and corded system telephones. Instead of a continuous display of search results, you have to first enter a series of characters when entering for a search. Search results can only be displayed after starting the search.
- Each individual LDAP-queried record is reserved for a definable time in a Forum 523/524 communications system cache (setting: **Delete cache** in minutes, 50 cache entries per company each with a separate time counter per entry). You should shorten this time (e. g. to 10 minutes) if, for example, the assignment of guests and room telephones in a hotel is managed via the LDAP server. If you set the **Delete cache** setting to 0 minutes the cache is not automatically deleted. Delete the cache as needed. Select the desired company entries on the **Telephony**: **Extended**: **Companies** page and click on the **LDAP Delete cache** button.

- Ideally, call numbers on the LDAP server are saved in the E.123 format beginning with a plus sign and country code. Furthermore, the bundles configured in the standard route of a company should have the setting **E.164 conversion** (see *E.164 conversion* starting on page 165). Special call numbers such as the emergency call number “110” should not be managed via LDAP, rather in the telephone book of the communications system.
- A call number queried via LDAP is adjusted before use. First of all, the characters in brackets beginning with “0” are deleted such as, e.g. “+49(030)1234”. Then all characters are deleted that are not numbers or plus signs.

  If the resulting call number is in the E.123 format, the initial plus sign is replaced with “00”. In addition, the seizure code of the company standard route is prefixed and (if configured, see *E.164 conversion* starting on page 165) the own prefix and the own base call number is deleted as necessary.

  If the resulting call number is not in the E.123 format the **Phone No.** option on the **Telephony**: **Extended**: **Companies** page determines how it is processed further:

  – Option **Phone No.** is set to **transparent**: The call number queried from the LDAP server is used without any further change.

  – Option **Phone No.** is set to **convert**: Select this option if the LDAP server provides call numbers without seizure codes. In addition, call numbers beginning with “0” have the seizure code of the company standard route prefixed and (if configured, see *E.164 conversion* starting on page 165) the own prefix and the own base call number is deleted as necessary.

  – Option **Phone No.** is set to **convert with trunc code**: Select this option if the LDAP server provides call numbers with seizure codes for the company standard route. In addition, for call numbers beginning with the seizure code of the company standard route (if configured, see *E.164 conversion* starting on page 165) the own prefix and the own base call number is deleted as necessary.

- Via the LDAP server you can also query another communications system from the family /Forum 500/ Forum 5050. If, for example, you connect multiple communications systems into a network (see *PBX Networking* starting on page 152) you can then manage call number information on a central communications system. Please note that information queried via LDAP cannot be forwarded via the LDAP server of the same communications system.

Use the Forum Name program for more advanced LDAP configuration, for connecting additional data sources or for additional call number conversions. After authorisation via a license key, the Forum Name program offers a LDAP service which can be queried by the Forum 523/524 communications system. The Forum Name program provides you with the following features:

- Bundling multiple data sources: Querying one or multiple LDAP servers and file sources in the CSV and LDIF formats.
- Rule-based call-number conversion, separate configurable rules for the call-numbers-to-name search and for conversion of call numbers for outgoing calls.

SNMP Interface

Components connected to the network are monitored using network management software in many companies. The management software ("SNMP-Manager") communicates with the network components via SNMP (Simple Network Management Protocol).

You can integrate the Forum 523/524 communications system into the SNMP-based network management. There is a Windows service program for this ("SNMP Agent"). The SNMP Agent communicates with the Forum 523/524 communications system via the CSTA interface. The status information queried via the CSTA interface is forwarded by the SNMP Agent to the network management software.

Note: *The SNMP Agent uses the CSTA interface of the Forum 523/524 communications system in an operating mode so designated. This is why no additional CSTA licence key is required for operation.*

The SNMP Agent supports SNMP Traps after SNMP version 2. In case the SNMP Agent cannot be accessed, the Forum 523/524 communications system stores up to 100 error or status messages. Also note that there is a SNMP description file (MIB, Management Information Base) for the Forum 523/524 communications system which can be found on the product CD .

Installing SNMP Agent

Install the SNMP Agent on a Windows PC. Select a PC which is usually in continuous use, e. g. a Windows server.

1. Log on to the PC as a user with administrator rights.
2. Start the installation program for the SNMP Agent (file name: “Forum5000_Setup.exe”). The program is on the product CD in the **...\Forum** directory.
3. Follow the steps of the installation assistant. In the **Choose Components** step deactivate all components except for the **SNMP Agent** component.
4. As you continue with the installation assistant the **IP address** step is now displayed. Enter the IP address of the Forum 523/524 communications system.
5. Terminate the installation assistant with the **Next**, **Install** and **Finish** buttons.

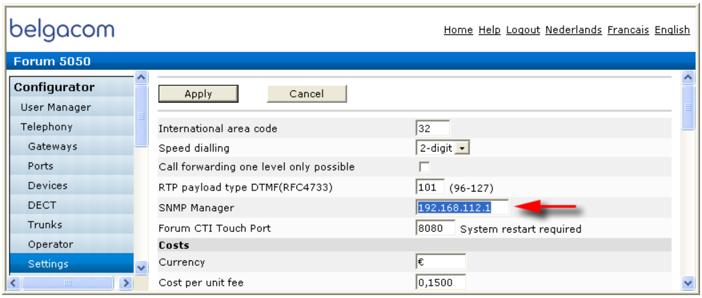
During installation a Windows service “Forum 5000 Supervisor Service” is configured. This service starts the SNMP Agent. This means that the SNMP Agent is automatically started with Windows and runs in the background, even when no Windows user is logged on.

Configuring SNMP

The SNMP Agent requires the IP address of the SNMP Manager for operation. System messages in the form of SNMP Traps are sent to this IP address. You manage the IP address of the SNMP Manager in the Forum 523/524 communications system configurator.

1. Start the web console of the Forum 523/524 communications system.

2. Navigate to the **Telephony** page in the **Configurator**:**Settings**. Click on the **Change** button.



Telephony: Settings: SNMP

3. Enter the IP address of the SNMP Manager into the **SNMP Manager** input field.

4. Confirm with **Apply**.

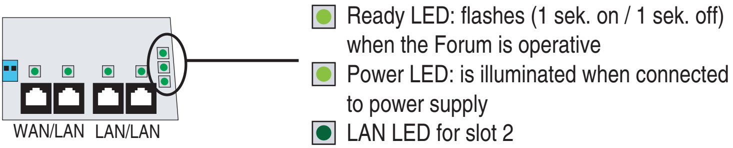
The IP address of the SNMP Manager is made available to the SNMP Agent in a configuration file via the TFTP server of the Forum 523/524 communications system.

Frequently Asked Questions

This chapter provides tips and information on how to deal with any malfunctions or faults you may experience with the Forum 523/524.

Please note: *Repairs to the Forum 523/524 should only be carried out by qualified personnel.*

The following LEDs indicate that the Forum 523/524 is ready for operation:



Position of LEDs on the Forum 523/524
The WAN/LAN LEDs are illuminated, if a device is connected

General/Hardware

Question: The Forum 523/524 is not functioning.

Make sure the mains plug is properly connected.

Plug another device into the mains socket to check whether there is any voltage.

Question: The mains plug is connected, the mains socket is supplying output, but the Forum 523/524 still does not function.

DANGER! This device contains hazardous voltages. To make the system dead, remove the power plug from the socket.

Open the housing (see *Opening and Closing Forum 523/524* starting on page 36). Is the power LED illuminated?

If not, contact your service centre or an authorised dealer. The AC adapter plug of the Forum 523/524 may be defective.

Question: After restarting the Forum 523/524, nothing is indicated on the displays of any connected terminals.

It takes a short while for the Forum 523/524 to start up.

After the restart, check whether the activity LED blinks at a rate of 10s / 1s. This indicates that the Forum 523/524 has started up correctly and is ready for operation. If the activity LED is not illuminated or the LED is dark after a short period of rhythmical blinking or the LED blinks 3 times fast in the meantime, the restart was not successful.

If the Forum 523/524 has not restarted properly, reset the Forum 523/524 to its original factory setting (refer to the chapter entitled *Resetting the System Data* starting on page 83).

Telephony

Question: It is not possible to make external calls.

Check the connection between the NTBA and the Forum 523/524.

In the **Configurator**, check whether the external S_0 ports are configured correctly (**Telephony: Ports: S_0** menu):

– Configuration of System- / Multi-terminal access OK?
– Port is connected to the NTBA?
– Faultless Cabling?
– Terminating resistors properly configured?

Check the status of the trunks. In the Configurator, open the **System Info**: **Telephony**: **Trunks** page. For the **Trunks** used for the “External trunk” route the **Status** indicator should display a small green hook symbol.

Question: The Forum 523/524 is connected to an NTBA with a multi-terminal configuration. Why is it not possible to establish external connections?

With the original factory setting, an additional external S_0 port is set for an NTBA in the communications system configuration; this additional port will be used first to seize a trunk line.

Deactivate the corresponding S_0 port in the **Configurator** (**Telephony: Ports: S_0** menu).

Question: One of the telephones is not functioning at all.

Make sure the telephone has been properly connected. Please read the explanations under *Port Assignment, Termination, Cable Length* starting on page 48.

Check also whether the appropriate port has been configured correctly in the **Configurator** (**Telephony: Ports** menu). For IP system telephones, verify if PoE is enabled in the **Network**: **LAN** menu. Please pay attention to the notes about the *Power Supply Unit* under *Forum Phone 515 / 525 / 535: Extensions and Accessories* starting on page 61.

Question: It is not possible to make external calls with one of the telephones.

Check whether a user is configured for the telephone. Otherwise the settings of the Guests user group are valid for the telephone. To standard, this user group has no external call authorisation.

Make sure the user configured for this telephone belongs to a user group with external line access (**Configurator**, **User Manager: User groups** menu).

Check also whether the internal call number of this telephone has been configured for incoming call distribution (**Configurator**, **Telephony: Call Distribution** menu).

Question: One of the features (e.g. call diversion) on one of the telephones cannot be used even though the feature has been configured in the **Configurator** of the Forum 523/524.

Make sure the user configured for this telephone belongs to a user group that has access to this feature (**Configurator**, **User Manager: User** and **User groups** menus). Some features cannot be used until the system PIN is changed.

Question: Nothing is indicated on the display of one of the connected ISDN telephones.

You have connected the ISDN telephone to an external S_0 port (RJ-45 socket). These ports are intended for connection to the NTBA only. Connect the telephone to an internal S_0 port (pressure terminal).

Question: Calls can be made but not received with one of the ISDN telephones.

The internal call number that has been configured for this ISDN telephone in the **Configurator** (**Telephony: Ports: S0** menu) must also be configured as an MSN on the ISDN telephone itself. For further information, refer to the User Guide of your ISDN telephone.

Question: An ISDN telephone always rings, if another telephone on the S_0 bus is being called.

This case also requires configuring the MSN on the ISDN telephone (see above answer).

Question: It is not possible to configure **Call Distribution: Outgoing** for multi-terminal access.

You have configured multi-terminal access and system access in parallel. All outgoing calls are therefore established via system access, and outgoing call distribution can be configured for system access only in the **Configurator** (**Telephony: Call Distribution** menu).

A specific MSN can be seized for individual calls by means of a code number procedure. For further information, refer to the "Forum 500 / Forum 5000 – Standard Terminals" user guide.

Question: What are some of the causes for problems when sending and/or receiving faxes?

In frequent cases, the reason may be found in a problem with the ISDN-L1 reference clock distribution. The L1 clock is delivered from the network provider. An unclean L1 clock distribution and the introduced signalling jitter is overheard by the human ear. Nevertheless, data and fax transmissions may be disturbed by the jitter. Please check, which ISDN lines will deliver the L1 clock. Details can be found under *L1 Clock* starting on page 155.

The fax data transfer possibly is routed via a compressing VoIP connection. Please verify, if the a/b port is configured with the "Fax" setting. For the VoIP connection, select a VoIP profile which includes the non-compressing G.711 codec.

PBX Networking

Question: Why is it not possible to answer calls from another communications system via the missed calls list?

You get a call e.g. via a direct Q.SIG connection. Which line seizure codes you will need to dial for the callback is not included automatically in the information transferred with Q.SIG. Therefore, you need to setup the callback in the bundle configuration.

- Call up the **Telephony**: **Trunks**: **Trunk group** page. Change the desired Q.SIG bundle and enter the line seizure code for the call-back in the **Prefix for source phone number at incoming internal** input field.

Question: Why is it not possible to reach a communications system indirectly via another communications system?

You have interconnected e.g. three communications system with two direct Q.SIG connections. While configuring the routes, you applied the default “Business” selection for the **Type** setting. Call-switching for internal calls is possible for the in-between communications system for internal routes only. Change the Type setting for all affected routes to “Internal”.

DECT

Question: The LED of the Forum base station is flashing, but none of the DECT devices is functioning.

Make sure the terminal setting for the corresponding U_{pn} port is set to an Forum base station (**Configurator**, **Telephony: Ports: U_{pn}** menu).

If multiple Forum base stations are installed, the blinking LED indicates that synchronisation is not finished.

Question: The LED of the Forum base station is continuously lit up, but one of the cordless DECT devices is indicating “No connection”.

You have not registered this DECT device. Configure a port in the **Configurator** and start the enrolment procedure (**Telephony: Devices: DECT Phones** menu).

Question: Is it possible to increase the time for the enrolment procedure?

You must manually enter the IPEI of the DECT device in the **Configurator**. The enrolment time is then increased to one hour (**Telephony: Devices: DECT Phones** menu).

Question: Another manufacturer’s DECT device is not functioning.

Check whether the DECT device supports the DECT GAP standard. In the **Configurator**, also make sure **GAP** is set for this DECT device (**Telephony: Devices: DECT Phones** menu).

Question: The startup procedure of the DECT base stations take a long time? What is the reason?

This behaviour may indicate a problem with the reference clock. Refer also to *What are some of the causes for problems when sending and/or receiving faxes?* starting on page 223.

LAN

Question: Why is it not possible to establish a network connection with the Forum 523/524?

Check whether the LEDs for the switch and the PC’s network card are indicating a connection.

Check the LEDs for the LAN functions of the Forum 523/524. A green LED above each LAN port indicates whether the network cable has been properly connected. A blinking green LED indicates whether there is any network traffic on the line.

To check whether there is a network connection with your Forum 523/524, enter the “ping IP address” command in “Run” in the Windows Start menu (e.g. ping 192.168.99.254).

Question: How can I determine the IP address of the Forum 523/524?

To find out what the IP address is, enter the code number *182 on one of the connected system telephones.

The code-number procedure *183 also displays the network mask.

Question: The network connection is functioning, but nothing is displayed in the browser.

Enter the complete IP address of the Forum 523/524 along with the protocol identifier, for example http://192.168.99.254/.

Check whether the browser has been configured for connection through a proxy server. If so, deactivate the “Connect through proxy server” setting.

Question: You have just configured the Forum 523/524 via the network. Why is it not possible now to establish a remote data transfer network connection?

The network card and the communication (remote data transfer) adapter cannot be run with the same routing setting. Deactivate the network card before connecting via the dial-up network.

Question: Our network has grown over time, with several segments connected by one central router. How can PCs from all segments connect to the Forum 523/524?

If several routers are configured for your network in different segments, you can enter extra static routes in the **Network: Routes** menu.

Question: In our network the Forum 523/524 dynamically issues the IP addresses by DHCP. Can I firmly assign the IP address for our internal server PCs (mail, Web)?

You need a static address assignment for these PCs. Make the appropriate host assignment entries in the Configurator (**Network**: **Hosts** menu). Create a static DHCP entry for each host assignment in the **Network**: **DHCP** menu. Activate “Dynamic and static address” for the DHCP server.

Internet

Question: I cannot access our company Web site.

Outside your system, your company Web site is accessed at “www.firm.com”, but in the **Configurator** you have entered “firm.com” as the domain. Your company’s site URL thus counts as an internal URL and can only be accessed by entering the direct IP address. If required, change the domain setting in the **Network**: **LAN** menu.

Question: Why do some Internet services not work even though they can be used when dialling in directly via a modem?

Some Internet services require an active connection coming from the Internet. But the configured filter rules prevent this. Plus, it is not possible to establish incoming Internet connections with the PCs directly owing to the network address translation process.

It is possible to redirect incoming connections in the **Configurator**, menu **Network**: **Port Forwarding**. You should secure the redirection target (PC or server) with a suitable firewall software.

Question: A SIP connection only passes unidirectional voice. What is the reason?

You did not use the Forum 523/524 as internet router or the STUN server of the SIP provider is unavailable. You need to activate the SIP support at your internet router, such as “SIP-ALG” or “Full Cone NAT” functions. You can also use the Forum 523/524 for internet access. Correct the STUN setting in the **Telephony**: **Trunks**: **SIP Provider** menu.

Question: Is it possible to use Q.SIG-IP connections via an Internet access with dynamic IP address?

Q.SIG-IP connections require a fixed IP assignment for technical and security reasons. Therefore you need an Internet access with a fixed IP address. It is possible to tunnel a Q.SIG-IP connection through a VPN connection. A VPN connection offers the possibility to determine the peer’s IP address with a DynDNS service during connection setup. VPN and DynDNS can be realized with external servers or routers also.

Technical Specifications

System data	
Mains power supply	230 V ~ 50 Hz
Rated power	205 VA
Safety class	2
Permissible temperatures, stationary, weatherproofed	+5 °C to +40 °C
Dimensions (W x H x D)	366 x 368 x 124 mm
Weight (system only)	2.0 kg
S_0 ports	
Euro ISDN external (S_0 external) for basic access, DSS1 protocol	1 x
Euro ISDN switchable (S_0 external / S_0 internal) for basic access, DSS1 protocol, or for ISDN terminals, DSS1 protocol	1 x with interface cards: up to 5 x; possible interface cards with S_0: – 4 x S_0 – 2 x S_0 and 6 x U_{pn} with DECT – 2 x S_0 and 6 x a/b
– Supply voltage	40 V ± 10%
– Supply power	3 VA for internal
– Range	150 m internal
U_{pn} ports	
for system telephones and DECT base stations	3 x to connect system terminals, DECT-enabled; with interface cards: up to 11 x; possible interface cards with U_{pn}: – 4 x U_{pn} with DECT – 8 x U_{pn} with DECT – 2 x S_0 and 6 x U_{pn} with DECT

– Supply voltage	40 V ± 10%
– Supply power	3 VA per U_{pn} bus
– Range	1,000 m
a/b ports	
for analogue terminals with pulse or DTMF dialling, flash duration of 60 to 310 ms	4 x with interface cards: up to 12 x; possible interface cards with a/b: – 4 x a/b – 8 x a/b – 2 x S_0 and 6 x a/b
– Supply voltage	40 V ± 10%
– Supply power	1.2 VA
– Feed current	25 mA
– Range	1,000 m

Actor	
Contact load of actor	2 A / 125 V
– Voltage range	$U_{\approx}$ = 5 V...30 V
Ethernet interfaces	
WAN	1 x 10/100 Base-T
LAN	3 x 10/100 Base-T
Slots	
– for interface cards	2 x Slot 1 for interface cards Slot 2 with connection to an ethernet switch for a Media Gateway Card

Note: *The online help provides an overview of the limits that should be observed when configuring the Forum 523/524.*

Environmental Information

- The equipment that you bought has required the extraction and use of natural resources for its production. It may content hazardous substances for the health and the environment.
- In order to avoid the dissemination of those substances in our environment and to diminish the pressure on the natural resources, we encourage you to use the appropriate take-back systems. Those systems will reuse or recycle most of the materials of your end life equipment in a sound way.
- The crossed-bin symbol invites you to use those systems.

- If you need more information on the collection, reuse and recycling systems, please contact your local or regional waste administration.
- You can also contact us for more information on the environmental performances of our products.

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Technical support for your Forum® telephone exchange

belgacom

User instructions

All the user instructions for our Forum® telephone exchanges are available on the included CD ROM, and on our Internet site at: www.belgacom.be/pabx.

Diagnosing the problem

In the event of a technical failure or problem, we request that you systematically carry out the following tests before contacting our technical support service.

To facilitate the diagnosis, please inform the helpdesk operator(1) of these test results.

Our helpdesk can solve certain problems remotely. This will avoid a technician's visit and reduce repair time considerably.

Problem with a telephone connected to your exchange

1. Reset the telephone by unplugging it and plugging it back in again.
2. Check the connections, cables and the various plugs. Try it out with the cables from another telephone that is working properly.
3. Test the defective telephone in another socket into which a functioning telephone is plugged.

Problem with a cordless DECT telephone

1. Remove and then reinsert the batteries.
2. Ensure that the charger is properly connected.
3. Test the DECT telephone near each of the antennas.

General external or internal communication problem

1. Check that the installation is always powered by 230V.
2. Reset the PABX:
   Switch off the 230V power supply for a few seconds.
   Disconnect the Forum's UPS backup power supply, too.
   Plug it back in and wait a few minutes for the system to start up again
   NB: You will not be able to use the system during this period
   (for up to 30 minutes).

If the problem persists, contact our technical support service.

(1) Contact our technical support service

If the problem persists after these few tests:
Consult our Internet site www.belgacom.be or contact our technical support 24/24 at the following numbers:

Technical support for your Forum® telephone exchange

- for SMEs and residential customers: 0800/55700
- for large companies 0800/55100

Change your configuration

To change your installation, add equipment or adapt the programming on your Forum®, contact our Customer Service Department at the following numbers:

- for SMEs and residential customers: 0800/55800
- for large companies 0800/55200

For more information:

- Please dial 0800 55 400
- Visit us in the Internet at www.belgacom.be/pabx
- Contact your Belgacom dealer

Subject to changes

Status 09.2011